The Principles of Electronic Communication

EB-551A-50 Instructor's Guide 595-5874-03

The Principles of Electronic Communication, Instructor's Guide
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INTRODUCTION

This Instructor's Guide accompanies the Heathkit Educational Systems course The Principles of Electronic Communications. The students will be reading the Textbook and performing the exercises in the Lab Book; the Instructor's Guide gives the answers to all the questions within these two books. This Instructor's Guide was developed to go along with the EB-551A Textbook, part number 595-5872 and subsequent versions, and the EB-551A-70 Lab Book, part number 595-5873.

To the greatest extent possible, this guide does not duplicate material in the Textbook or Lab Book. For example, answers to pertinent exercise procedure questions are presented in this guide in the context of the original question; however, those sections of the exercise which do not require an answer are not reproduced here.

This Instructor's Guide provides guidelines and information for all phases of the course. First, it lists the course prerequisites, objectives, and outline. Second, it provides a sample lesson plan, which outlines the recommended sequence for text and exercise completion. Third, it provides details and answers for the Lab Book exercises. Fourth, it provides the answers to the Self-Test Reviews within the Textbook. Finally, this guide provides Chapter and Final examinations (and their answers), allowing you to properly test your students.

COURSE PREREQUISITES

This course assumes that the student has already completed basic electronics training. The student should already understand basic circuitry, oscilloscope and multimeter operation, and calculations which involve DC and AC circuits. The Principles of Electronic Communications is designed to follow basic electronics.

COURSE OBJECTIVES

Upon completion of this course, the student should be able to:

1. List the frequency ranges of audio, voice frequencies, and the divisions of the electromagnetic spectrum.

2. Calculate power gains, losses and reference power in dB units, and convert these levels to voltage, current, and power ratios.

3. Identify common signal waveforms, and show how they are created by combining multiple frequency sine waves.

4. Specify the common forms of waveform distortion applied to signals as they pass through electronic circuits.

5. Determine the effects of selective filtering on the suppression and/or selection of specific harmonic signals.

6. Identify and calculate the intermodulation frequencies that result from processing two signals through non-linear circuits.

7. Identify the effects of phase and filtering on the generation and suppression of harmonic and intermod signals.

8. Explain the operating principals of input/output transducers.

9. Explain the characteristics of FETs and ICs.

COURSE OUTLINE

Introduction

Section 1 — Electrical Signals

Chapter 1 – Signal Characteristics
- Introduction
- Objectives
- Acoustic and Electric Audio Signals
- Telecommunications Signals
  - Electromagnetic Signals
  - Frequency Ranges
- Dimensions
  - Wavelength
- Phase
  - Frequency vs. Time
  - Time and Frequency of Electrical Networks
- Noise in Signals
- Summary
- Self-Test Review

Chapter 2 – Signal Power
- Introduction
- Objectives
- Sine Wave Signal Power
  - Practical Examples
- Power in Networks and Circuits
  - Power Factor
- Network Calculations
- Power Gains and Losses
- Summary
- Self-Test Review

Chapter 3 – Logarithmic Power Ratios
- Introduction
- Objectives
- Calculating Decibels
- Quick dB Conversions
  - Voltage (or Current) Ratios
- Referenced Absolute Power
  - Scaling Power in dBm
  - Practical Examples
  - Reverse Calculations
- Power Scaling with Prefixes
- Summary
- Self-Test Review

Chapter 4 – Power/Frequency Applications
- Introduction
- Objectives
- Acoustical Signal Power
- Frequency Response of Signals
- Frequency Response of Circuits
  - Practical Examples
  - Phase Relationships
- Signal-to-Noise Ratios
- Signal Power Transfer
  - Mismatch Loss
  - Transmission Lines
- Summary
- Self-Test Review

Section 2 — Waveforms And Harmonics

Chapter 5 – Frequencies Within Signal Waveforms
- Introduction
- Objectives
- The Ideal Sine Waveform
  - Non-Sine Waveforms
  - The Effects of Combining Signals
- Waveforms Containing Multiple Frequencies
  - Creating Waveforms
  - Communications Signals
- Summary
- Self-Test Review

Chapter 6 – Frequency and Power Distribution
- Introduction
- Objectives
- The Spectra of Complex Waveforms
  - Significant Power Levels
  - Frequency Distribution
- Bandwidth and Bandpass
- Harmonic Distortion
- Summary
- Self-Test Review

Chapter 7 – Filters
- Introduction
- Objectives
- Filter Characteristics
- Filter Types
- Equalizing Networks
- Summary
- Self-Test Review

Chapter 8 – Utilizing Waveforms and Harmonics
- Introduction
- Objectives
- Harmonic Suppression
  - Filtering
- Harmonic Generation
  - Harmonic Generators
- Harmonic Multipliers
- Summary
- Self-Test Review

Section 3 — Combining Signals

Chapter 9 – Multiple Signals in Linear Circuits
- Introduction
- Objectives
- Superposition
  - Superposition Calculations
  - Calculating Voltage and Frequency
- Linear Circuits
- Summary
- Self-Test Review

Chapter 10 – Multiple Signals in Non-Linear Circuits
- Introduction
- Objectives
- Intermodulation Distortion
- Frequency and Power Distribution of Intermod Signals
  - Intermod Frequencies
  - Intermod Distortion Factor
  - Bandwidth
- Summary
- Self-Test Review

Chapter 11 – Applications of Intermodulation Distortion
- Introduction
- Objectives
- Suppressing Harmonic and Intermod Signals
- Modulation
  - Amplitude Modulation
  - Sideband Power Distribution
  - Sideband Modulation
- Mixers
  - Frequency Conversion
  - Balanced and Unbalanced Mixers
- Summary
- Self-Test Review

Section 4 — Creating, Amplifying, and Reproducing Signals

Chapter 12 – Transducers in Audio and Telephone Systems
- Introduction
- Objectives
- Transducers
- Electrical Properties of Input Transducers
  - Sensitivity of Microphones
  - Frequency Response
  - Noise and Distortion
  - Microphone Impedance and Power Transfer
- Electrical Properties of Output Transducers
  - Efficiency
  - Frequency Response
  - Impedance, Power, Distribution
- Summary
- Self-Test Review

Chapter 13 – Audio Voltage and Power Amplifiers
- Introduction
- Objectives
- Transistor Amplifiers
- Combining Amplifier Circuits
  - Signal Levels and Distortion
  - Amplifying Low-Level Signals
  - The Complete Amplifier
- Integrated Circuit Audio Amplifiers
- Passive Components
- Summary
- Self-Test Review

LESSON PLAN

The general lesson plan is divided into four sections, corresponding to the four sections in the Textbook. Each section is divided into chapters, corresponding to the text prior to each Self-Test Review. It is up to you to personalize your lesson plan. Ample space is provided for detailed notes.

Each exercise is listed under "Laboratory Activity" in the General Lesson Plan, immediately following the Section in which the topic is covered. It is up to you to decide upon the best time to perform each exercise.

A Sample Lesson Plan follows, indicating how much material could be taught each week for a 6-week course. This sample is by no means a hard and fast rule, only a guideline.

SAMPLE LESSON PLAN

	Section	Chapter	Subject
Week One:	1	1	Signal Characteristics
	1	2	Signal Power
		*L	Introduction to Trainer and Lab Equipment
	1	3	Logarithmic Power Ratios
	1	4	Power/Frequency Applications
		L	Lab Exercise 1
	1	*E	Section Review and Exam
Week Two:	2	5	Frequencies Within Signal Waveforms
		L	Lab Exercise 2
	2	6	Frequency and Power Distribution
	2	7	Filters
		L	Lab Exercise 3
		L	Lab Exercise 4
	2	8	Utilizing Waveforms and Harmonics
		L	Lab Exercise 5
	2	E	Section Review and Exam

* Segment L indicates Lab time.
Segment E indicates Review and Exam Time.

	Section	Chapter	Subject
Week Three:	3	9	Multiple Signals in Linear Circuits
	3	10	Multiple Signals in Non-Linear Circuits
		L	Lab Exercise 6
	3	11	Applications of Intermod Distortion
		L	Lab Exercise 7
			Section Review and Exam
Week Four:	4	12	Transducers in Audio and Telephone Systems (Part 1)
		L	Lab Exercise 8
	4	12	Transducers in Audio and Telephone Systems (Part 2)
Week Five:	4	2	Audio Voltage and Power Amplifiers (Part 1)
		L	Lab Exercise 9
	4	13	Audio Voltage and Power Amplifiers (Part 2)
		L	Lab Exercise 10
	4	E	Section Review and Exam
Week Six:			Course Review
			Lab Equipment Turn-In
			Final Examination

GENERAL LESSON PLAN

SECTION 1 ELECTRICAL SIGNALS
Chapter 1 — Signal Characteristics

[Time (hours):_____]

Objectives Covered:

1. Define the properties of signals in both acoustic and electrical form.

2. Describe and dimension signal frequency, wavelength, and phase.

3. Apply the terms and formulas of basic AC to electrical signals.

4. List the frequency ranges of audio, voice frequencies, and the divisions of the electromagnetic spectrum.

5. Calculate the relationships of time and frequency, phase and distance, and phase and time.

Topics Covered:

- Class discussion about telecommunications as a career field, and identification of the subjects in this text that will prepare students to understand the technical aspects of this field.

- Identify the distinctions between acoustic and electrical signals, and offer examples of how both of these signals are used in telephone practice.

- Continue with the identification of telecommunication signals, with the introduction of the concept of frequency, the audio spectrum, and the electromagnetic spectrum.

- Review the units by which signal frequency and wavelength are dimensioned; including the relationships of wavelength, frequency, time, and phase.

- Describe the relationships between time, frequency, and resonance, as created in electrical LC networks.

- Conclude with an explanation of noise and signal-to-noise ratio.

Assignment:

Self-Test Review

INSTRUCTOR NOTES

SECTION 1 ELECTRICAL SIGNALS
Chapter 2 — Signal Power

[Time (hours):_____]

Objectives Covered:

1. Apply the terms and formulas of basic AC to electrical signals.

2. Use basic terminology to describe and dimension signal power.

Topics Covered:

- Identify the four factors of electrical energy (EMF, resistance, current, and power), emphasizing power as the factor that defines electrical work.

- Continue with the importance of power levels as the major dimension (along with frequency) used to define signals; and emphasize the distinctions between the DC power used to operate electronic devices, and the signal power which is processed by electronic devices.

- Review the common units in which power is defined, and the Ohm's law relationships from which power is computed, including the more useful rms and p-p formulas, such as:

$$P = \left[\left(V_{rms} \right)^2 \div R \right] = \left[\left(V_{p\text{-}p} \right)^2 \div 8R \right]$$

$$V_{rms} = \sqrt{(PR)}$$

- Introduce and review examples of signal power applied to common electrical devices from your experience, and familiar to the students. Applying signal power to an 8-ohm loudspeaker is a good example; e.g. what signal voltage will produce 1 watt, or 50 watts across the speakers of a home or car stereo system.

Assignment:

Self-Test Review

INSTRUCTOR NOTES

SECTION 1 ELECTRICAL SIGNALS
Chapter 3—Logarithmic Power Ratios

[Time (hours):_____]

Objectives Covered:

1. Calculate power gains and losses in dB units, and convert these levels to voltage, current, and power ratios.

2. Calculate referenced power levels in dBm, and convert these levels to voltage, current, and power ratings.

Topics Covered:

- Specify the logarithmic relationships of power in physical phenomena, including acoustic and electrical signals. Define the decibel as a measure of signal power ratio.

- Continue with the techniques used to calculate and convert decibels to power ratios, and vice versa; then, apply these techniques to calculate decibels and power levels using examples found in actual telecommunication practice.

- Review the dBm as a measure of how signal power is scaled in reference to a 1 mW standard; then describe other similar standards; e.g., dBf, dBrnC, etc. Make distinctions between gain and loss measured in dB, signal power in dBm, and acoustic power in dB.

- Apply power scaling to a sequence of test points along a communication circuit that has both gains and losses, with referenced signal power measured at the circuit's input and output terminals. By example, calculate the power, in dBm and mW, at each of the interconnect points of the circuit using the Scaled Power Ruler Line of Figure 3-5 in the Textbook.

- Conclude with a demonstration of power measurements (in mW) made at different points along a circuit; then, convert these measurements to dBm, and finally calculate signal gain and loss between points in the circuit where signal power is measured.

Assignment:

Self-Test Review

INSTRUCTOR NOTES

SECTION 1 ELECTRICAL SIGNALS
Chapter 4—Power/Frequency Applications

[Time (hours):_____]

Objectives Covered:

1. Define the concepts of frequency response, power versus frequency, to telephone and audio equipment.

2. State the signal-to-noise ratios required for reliable communication with telephone and audio equipment.

3. Describe and dimension the characteristics of transmission lines and cables used for the efficient transfer of signal power.

Topics Covered:

- Relate the sound pressure levels of acoustical signals that are measured in dB to the loudness of common sounds familiar to students.

- Describe the general frequency response of human hearing using the Fletcher-Munsen curves as a reference. Include how we hear loud sounds more uniformly than we hear soft sounds, and how we have standardized our reference at 1000 Hz for the loudness of sound signals.

- Review the process by which frequency response measurements are made and documented. Then, apply the measurement process to an LCR network, such as a filter, to demonstrate a change in signal power at the filter's output over a range of frequencies.

- Identify the terminology of frequency and phase response as applied to filters; including the concepts related to the terms "corner frequency" and "pole." Continue by defining the character of frequency response slopes specified in dB-per-range of frequencies; e.g., 6 dB/octave, and 20 dB/decade.

- Continue with an introduction to noise as a random-frequency signal distributed across a range of frequencies, and signal-to-noise ratio as the dB relationship between two power levels; then, apply the concept of S/N to the clarity of communication signals.

- Conclude with a detailed discussion of the power losses that occur along transmission lines and circuits due to mismatched impedances; then, progress through the procedures for calculating mismatch loss in dB.

Assignment:

Self-Test Review

Section One Laboratory Activity:

- Review the operation of all laboratory equipment, the Heathkit ETW-5000 Trainer, and the ETB-551A Exercise Module.

- Exercise 1: Generating and Measuring Sine Wave Signals

Section One Examination

Give a comprehensive review of exam upon its completion.

INSTRUCTOR NOTES

SECTION 2 WAVEFORMS AND HARMONICS
Chapter 5—Frequencies Within Signal Waveforms
[Time (hours):_____]

Objectives Covered:

1. Define the fundamental and harmonic frequencies of electrical signals.

2. Identify common signal waveforms, and how they can be created by combining multiple frequency sine waves.

Topics Covered:

- Describe the sine wave in reference to the degrees of rotation, relating the amplitude of the voltage to phase angles from 0° to 90°.

- Review the concepts of superposition and the effects of combining signals of frequencies that are multiples of the fundamental; then, define the resulting character of the waveform in relation to the frequencies within that waveform. Relate this process to musical synthesizers that create an extensive variety of waveforms by superimposing multiple signal frequencies.

- Reverse the superposition process and describe how different waveforms contain multiple frequencies, and that the character of these waveforms defines each of the frequencies contained within them. Apply this concept to the common waveforms used in communication practice.

- Introduce the terminology relating to fundamental and harmonic frequencies, and define by example, the fundamental and harmonic frequencies contained within a non-sine waveform.

- Describe the frequency domain and the instruments, such as the spectrum analyzer, that are used to measure and display the frequencies and power levels of signals containing harmonics. Make comparisons and distinctions between the frequency-domain spectrum analyzer and the time-domain oscilloscope.

- Conclude with a description of common communication signals and their waveforms that include harmonics; then, relate a circuit's frequency response to the number of harmonics within a signal that will pass through a circuit.

Assignment:

Self-Test Review

INSTRUCTOR NOTES

SECTION 2 WAVEFORMS AND HARMONICS
Chapter 6—Frequency and Power Distribution

[Time (hours):_____]

Objectives Covered:

1. Define the harmonic structure of common signal waveforms, and identify their frequency and power distribution.

2. Specify the common forms of waveform distortion applied to signals that pass through electronic circuits.

3. Specify the effects of circuit bandpass and frequency response on signal waveforms and their harmonic structure.

Topics Covered:

- Introduce the spectra of common complex signal waveforms with a demonstration of both spectrum analyzer and oscilloscope displays, emphasizing both the frequency and power levels (including dB reference to the level of the fundamental frequency) of harmonic signals, as indicated by the spectrum analyzer.

- Apply the concepts of spectral distribution to the calculation of circuit bandpass required to retain all of the harmonics within a waveform. Continue with calculations that show how limiting circuit bandpass (through filtering) will limit the spectral distribution and reduce the harmonics of complex signal waveforms, thus simplifying the signals.

- Introduce and define the types and characteristics of popular filters; then, proceed to calculate the relationships between slope, bandpass, and Q. Apply the concept of filtering to the equalizing networks used in telephone practice.

- Introduce the concept of harmonic distortion, wherein harmonic frequencies are created through the distortion of a sine wave signal.

- Continue with a review of procedures for the calculation of harmonic distortion. Compare the power level of the fundamental with the power levels of the harmonic signals; then, describe the applications of harmonic distortion used in communication practice.

Assignment:

Self-Test Review

INSTRUCTOR NOTES

SECTION 2 WAVEFORMS AND HARMONICS
Chapter 7—Filters

[Time (hours):_____]

Objectives Covered:

1. Determine the effects of selective filtering on the suppression and/or selection of specific harmonic signals.

2. Relate the concepts of harmonic distortion and frequency generation to telephone, audio, and telecommunication systems.

Topics Covered:

- Introduce the signal requirements of communication circuits and describe how harmonics are suppressed by limiting circuit bandpass with filters.

- Discuss the four general categories of filters: low-pass, high-pass, bandpass, and bandstop.

- Describe how equalization networks are used in communications circuits.

Assignment:

Self-Test Review

INSTRUCTOR NOTES

SECTION 2 WAVEFORMS AND HARMONICS
Chapter 8—Utilizing Waveforms and Harmonics

[Time (hours):_____]

Objectives Covered:

1. Relate the results of waveform distortion to the generation of harmonic frequency signals.

2. Apply the concepts of harmonic generation and frequency response to the analysis of electronic comb and marker signal sources.

3. Relate the concepts of harmonic distortion and frequency generation to telephone, audio, and telecommunication systems.

Topics Covered:

- Introduce the contrary requirements of communication circuits and systems to create and/or maintain harmonics within their bandpass.

- Describe harmonic generators (harmonic multipliers) and their use in creating frequencies which are multiples of a stable fundamental frequency; then, continue with specification of the harmonic signal power levels available from specific waveforms.

- Detail the operation of the basic FDM (frequency division multiplex) system that uses harmonic generation to assure the synchronous operation of the multiple voice channels it processes.

- Demonstrate the calculation of efficiency and frequency range in harmonic generators and multipliers, including doublers and triplers.

Assignment:

Self-Test Review

Section Two Laboratory Activity:

- Exercise 2: Measuring Power Gain and Loss
- Exercise 3: Measuring Signal Phase and Phase Shift
- Exercise 4: Frequency Response Measurements
- Exercise 5: Circuit Effects on Voice Signals

Section Two Examination

Give a comprehensive review of the exam upon its completion.

INSTRUCTOR NOTES

SECTION 3 COMBINING SIGNALS
Chapter 9—Multiple Signals in Linear Circuits

[Time (hours):_____]

Objectives Covered:

1. Identify and distinguish between linear and non-linear circuits that process two (or more) signals.

2. Calculate the effects of linear mixing (adding or summing) of two signals through a single circuit using superposition techniques.

3. Determine and identify the waveforms that result from the adding, or linear mixing, of signals.

Topics Covered:

- Describe the results of superposition as it applies to the combination of two signals within a linear circuit.

- Calculate the voltage and frequency of the signals within linearly combined waveforms.

Assignment:

Self-Test Review

INSTRUCTOR NOTES

SECTION 3 COMBINING SIGNALS
Chapter 10—Multiple Signals in Non-Linear Circuits
[Time (hours):_____]

Objectives Covered:

1. Determine and identify the waveforms that will result from the processing of two signals through non-linear circuits.

2. Determine and identify the intermodulation frequencies that result from processing two signals through non-linear circuits.

3. Calculate the percentage factors of intermodulation and harmonic distortion in circuits and signals.

4. Calculate the power distribution of intermodulation signals over their range of frequencies.

Topics Covered:

- Describe the result of combining two signals in a non-linear device.

- Identify modulation and intermodulation distortion as a result of one signal affecting the waveform of another; then, identify the sum and difference frequencies present within the waveform of a modulated signal.

- Review the process of calculating the intermodulation and harmonic frequencies produced when two signals are applied to a non-linear circuit. Then, specify the bandwidth of these signals and the circuit bandpass required to retain all of these frequencies.

- Discuss intermodulation distortion, how it is measured, and how to calculate %IMD.

Assignment:

Self-Test Review

INSTRUCTOR NOTES

SECTION 3 COMBINING SIGNALS
Chapter 11—Applications of Intermodulation Distortion
[Time (hours):_____]

Objectives Covered:

1. Specify the effects of circuit bandpass and frequency response upon the signal waveforms and intermodulation frequencies.

2. Determine the effects of phase upon the generation and suppression of harmonic and intermod signals.

3. Determine the effects of filtering upon the generation, suppression, and/or selection of harmonic and intermod signals.

4. Apply the concepts of intermod products to the modulation of signals in communication systems.

5. Apply the concepts of intermod products to the frequency conversion of signals in communication systems.

Topics Covered:

- Introduce the requirements for suppressing harmonics and intermod signals in communication circuits and systems by limiting circuit bandpass, applying phase cancellation, and filtering.

- Introduce the concepts of the modulation process which purposely uses intermodulation distortion; wherein, information signals are applied to carrier signals.

- Define the terminology of the modulation process, including: carrier and sideband frequencies, modulation index, and the modulation forms—DSB-AM, DSB-SC, and SSB.

- Review the use of filters in selecting and/or suppressing specific intermod signals, and describe the process of phase cancellation using the balanced modulation circuit to suppress specific modulation or intermod signals; e.g., how the carrier is phase suppressed to obtain DSB-SC and SSB.

- Demonstrate the techniques used to calculate sideband frequencies and power levels in AM, DSB-SC, and SSB circuits. Emphasize the distinctions between the power and voltage levels of sideband signals, and the total sideband power as the sum of (or twice) each sideband signal power level.

- Introduce the mixer as a form of AM modulator that is distinct in function from the modulator, and specify the use of the mixer in frequency conversion processes critical to communication systems. Clarify the distinctions between unbalanced, single-balanced, and double-balanced mixers.

Assignment:

Self-Test Review

Section Three Laboratory Activity:

- Exercise 6: Measuring Signal Waveforms and Harmonics
- Exercise 7: Applications of Harmonic Signals

Section Three Examination

Give a comprehensive review of the exam upon its completion.

INSTRUCTOR NOTES

SECTION 4 CREATING, AMPLIFYING, AND REPRODUCING SIGNALS

Chapter 12 — Transducers in Audio and Telephone Systems

[Time (hours):_____]

Objectives Covered:

1. Identify the electrical principles upon which popular input and output transducers operate.

2. Identify the acoustical and electrical properties of popular input transducers such as microphones and telephone transmitters.

3. Identify the acoustical and electrical properties of popular output transducers such as loudspeakers and telephone receivers.

4. Determine the efficiency of transducers from measured specifications, and convert dB SPL to electrical signal power.

5. Determine the electrical requirements of gain and power level for amplifier systems using input and output transducers.

Topics Covered:

- Introduce transducers as the elements in the telephone handset that convert acoustic signals to electrical signals, and vice versa. Identify the types of transducers used in microphones and speakers, and in telephone transmitters and receivers.

- Define the important electrical properties of microphones (telephone transmitters), including: sensitivity, frequency response, impedance, noise, and distortion.

- Define the important electrical properties of loudspeakers (telephone receivers), including: sensitivity (efficiency), frequency response, impedance, power, and distortion.

- Demonstrate the procedures for calculating the sensitivity of microphones, and converting their specifications to voltage levels; then, demonstrate the application of microphone data to determine impedance and power transfer relationships.

- Demonstrate the procedures for calculating the efficiency of speakers, and determining required signal power levels versus acoustic power.

Assignment:

Self-Test Review

INSTRUCTOR NOTES

SECTION 4 CREATING, AMPLIFYING, AND REPRODUCING SIGNALS

Chapter 13 — Audio Voltage and Power Amplifiers

[Time (hours):_____]

Objectives Covered:

1. Identify the distinctions used to classify voltage amplifiers versus power amplifiers in electronic circuitry.

2. Identify and categorize the dominant electron devices used to amplify signals, including bipolars, FETs, and ICs.

3. Identify and categorize the dominant circuit configurations in which bipolar, FET, and IC devices are placed for signal amplification.

4. Determine the operating characteristics of amplifier devices and circuits, including gain, impedance, and power levels.

5. Determine the effects of amplifier circuits upon the frequency response, noise, and distortion of signals passing through them.

Topics Covered:

- NOTE — We assume that the students have had a course in basic electronic devices and circuits for prior exposure to the content of this segment of Section Four.

- Review the properties of electronic amplifiers that are critical to circuit performance.

- Review the configurations of single-stage transistor amplifier circuit connections, their power requirements, and voltage and power gain performance.

- Describe the multi-stage transistor connections typical of those found within IC devices; including the discrete complementary-symmetry output driver, and the differential input voltage amplifier.

- Describe the structure of the IC operational amplifier, and how it consists of differential input and complementary output stages; then, apply this circuit to the amplification of voice signals used in telephone practice.

Assignment:

Self-Test Review

Section Four Laboratory Activity:

- Exercise 8: Combining Signals—Intermodulation Distortion
- Exercise 9: Application of Intermodulation Signals
- Exercise 10: Transducer Specifications

Section Four Examination

Give a comprehensive review of the exam upon its completion.

INSTRUCTOR NOTES

EXERCISE INTRODUCTION

The laboratory exercises are intended to teach **understanding** — the understanding of what, why, and how. Specifically, the goals of the activities intend to teach the following:

1. The application of test equipment and procedures to measure telecommunication signals and circuits.

2. The interpretation of measured data to determine the functions and operation of telecommunication circuits.

3. The analysis of measured data to better understand the theoretical basis of communication signals, circuits, and systems.

The Laboratory Workbook is designed to be used in a classroom environment which has strong instructor support. The application of test equipment and procedures must be demonstrated to students prior to their lab work. Even with a thorough introductory demonstration, students will still require assistance while they manage the measurements and procedures.

To assure instructor support, there are check-off boxes for the instructor's initial at critical points in the exercise procedures. Students should demonstrate both their understanding of the measurement techniques, and the implications of the measured data, to obtain an approval mark from the instructor. There should also be some discourse (chit-chat) between the instructor and student at these points to elaborate the learning process. Always keep in mind the goal of **understanding** on the part of the student. To be a success in a technical career field, students need to know what they are measuring, why they are measuring it, and how to make the measurement. The instructor can assure better understanding through discussions and individual demonstrations with students during the lab procedures.

The ideal weekly lab session may be three to four hours in duration, with one to two hours scheduled for instructor demonstration, and at least two hours scheduled for student laboratory activity. Since all students don't work at the same pace, however, it is likely that additional open lab time may be required for some students to complete the assigned exercises. There are a variety of alternative patterns of structure for the lab, but there should always be time scheduled for the instructor demonstration.

REQUIRED EQUIPMENT

The EB-551A course has been designed around the Heathkit ETW-5000 Telecommunications Trainer with ETW-3567 Backpack and ETB-551A Exercise Module. Students can complete all of the exercises in the Laboratory Workbook with only one additional piece of test equipment — an oscilloscope.

The ETW-5000/ETW-3567 contain the functions of many separate interconnected pieces of test equipment which makes lab work very time-efficient. The structure of the ETW-5000/ETW-3567 also emphasizes the systems test set concept of test equipment, typical of what is used in the telecommunications industry. The ETW-5000 includes a 5-waveform function generator that can be switched to modulate a 5-mode carrier generator, a frequency counter, wideband amplifier, attenuator, and power meter. The ETW-3567 includes a digital multimeter and interface access to the slide-in circuit board module. Both units include power supplies and loudspeakers.

The specific requirements for an oscilloscope to use with this lab course include dual-trace operation, with X-Y capability, 20 MHz vertical bandwidth, and 5 mV/div vertical sensitivity. Two ×1 probes with hook-on and BNC connectors are also required.

ETB-551A EXERCISE MODULE DESCRIPTION

The ETB-551A Exercise Module slides into the rails and interface connectors of the ETW-3567 backpack for operation with the test system in performing the laboratory exercises.

The ETB-551A module includes the circuitry to simulate the functions of an operational telephone central office, as well as line amplifiers, LC and tunable filters, and a balanced modulator. The filters are used to analyze the spectral distribution of harmonic and intermodulation signals, as well as function in performing standard transmission system measurements. Two electronic telephone sets connect to this board to simulate a complete operational telephone communications system.

To perform the exercises, the circuits are interconnected on the ETB-551A circuit board module with wires that easily plug onto circuit board pins.

EXERCISE ANSWERS

EXERCISE 1

Generating and Measuring Sine Wave Signals

6. Output ≈ **0.23** Vrms and **0.67** Vp-p

7. Output ≈ **0.088** mW

8. Output ≈ **–10** dBm

   Is this dBm value equal to the dBm value used to initially set the signal?

   yes

9. –8 dBm/600 ohms **0.3** Vrms **0.84** Vp-p
   –13 dBm/600 ohms **0.17** Vrms **0.48** Vp-p
   –17 dBm/600 ohms **0.11** Vrms **0.31** Vp-p

10. –7 dBm/600 ohms **0.35** Vrms **0.99** Vp-p
    –3 dBm/600 ohms **0.54** Vrms **1.53** Vp-p
    0 dBm/600 ohms **0.78** Vrms **2.20** Vp-p
    +2 dBm/600 ohms **0.98** Vrms **2.77** Vp-p

    Do the rms and p-p voltages of the signal change in the same direction as the change in dBm?

    yes

    Are the changes to the rms and p-p voltages proportional to the amount of change in dBm?

    no

21.

Hz	10k	11k	12k	13k	14k	15k	16k	17k	18k	19k	20k
dBm	0	+1	+1	+3	+6	+9	N/A	N/A	N/A	N/A	N/A

Figure 1-1

(**Note To The Instructor:** The results for Figure 1-1 through Figure 1-3 depend upon the individual assessments of the student. Each set of values will be different. However, the sample figures provided should give an indication of general results.)

23.

Hz	1000	900	800	700	600	500	400	300	200
dBm	0	0	0	+4	+6	+10	+10	+10	+12

Figure 1-2

28.

Hz	200	180	160	140	120	100	80	60	40	20
dBm	0	0	+1	+1	+1	+3	N/A	N/A	N/A	N/A

Figure 1-3

INSTRUCTOR NOTES

EXERCISE 2

Measuring Power Gain and Loss

4. Output ≈ **0.77** Vrms and **2.1** Vp-p

7. DETECTOR ≈ **+0.92** VDC

8. What do you suppose will happen to the power level of the signal when one of the ATTENUATION switches is placed in the circuit?

It should decrease.

9. Output ≈ **–20** dBm

10. Output ≈ **0.07** Vrms and **2.1** Vp-p

11. DETECTOR ≈ **0.095** VDC

12. Output dBm ≈ **–20** dBm

    Output Vrms ≈ **–0.7** Vrms

    Output Vp-p ≈ **0** Vp-p

    DETECTOR VDC ≈ **–0.825** VDC

17. ATT. OUT ≈ **0.21** Vp-p

18. What is the total amount of attenuation presently in the circuit?

Total attenuation = **–40** dB

Considering the range of the Meter, how could you verify that the total attenuation is actually applied to the circuit?

The voltage ratios between TP1014 and TP1011 should be 100 to 1.

19. ATT. OUT ≈ **0.021** Vp-p

What is the mathematical relationship between this p-p voltage and the p-p voltage at TP1014 (Channel 1)?

100:1

Based upon this mathematical relationship, is the total attenuation you found in Step 18 applied to the circuit?

yes

20. Output ≈ **0.007 Vrms**

Does the relationship between attenuated and unattenuated p-p voltages also exist for rms voltages?

yes

22. Are you able to make an accurate measurement at –80 dB?

no

Are you able to measure this –80 dB signal with the Meter?

no

25. Output (0 dBm) ≈ **0.77** Vrms

26. Output (0 dBm) ≈ **2.1** Vp-p

27. Output (–20 dBm) ≈ **0.077** Vrms

28. ATT. OUT (–20 dB) ≈ **0.21** Vp-p

32. Output ≈ **0.76** Vrms and **2.15** Vp-p

Based upon these measurements and the measurements taken in Steps 25 through 28, what is the power level of the signal at TP1010?

TP1010 ≈ **0** dBm

33. Based upon your answer in Step 32, how much gain has the amplifier introduced into the circuit?

+20 dB

Output ≈ **0** dBm

34. What is the power level of the FUNCTION GENERATOR output with this decrease?

–40 dBm

35. AMP. OUT ≈ **2.1** Vp-p

Based upon this measurement and the measurement taken in Step 34, what is the power level of the signal at TP1010?

0 dBm

36. Based upon your answer in Step 35, how much gain has the amplifier introduced into the circuit?

+40 dB

Output ≈ **0** dBm

INSTRUCTOR NOTES

EXERCISE 3

Measuring Signal Phase and Phase Shift

9. + Phase shift ≈ **81**° and ≈ **0.45** π rads

   – Phase shift ≈ **81**° and ≈ **0.45** π rads

10.

FREQUENCY	TIME/DIV
125 Hz	1 ms
2.5 kHz	50 µs
62.5 kHz	2 µs

Table 3-1

19. Output (Channel 2) ≈ **1.4** Vp-p

$$\frac{\text{Output p–p}}{\text{Input p–p}} \approx \mathbf{0.70}$$

21. Phase shift ≈ **–58.5**°

22. Voltage coefficient ≈ **0.40**

23. Phase shift ≈ **–67.5**°

24. Voltage coefficient ≈ **0.390**

30. Input ≈ **2.6** Vp-p

    Output ≈ **2.5** Vp-p

    Phase shift ≈ **–9**°

31. Voltage coefficient ≈ **0.96**

32. (**NOTE TO THE INSTRUCTOR:** The values in Table 3-2 are approximations only. The values your students arrive at may differ, but there should be a decrease in the voltage coefficient as the frequency increases.)

FREQUENCY (Hz)	TIME/DIV	INPUT P-P	OUTPUT P-P	VOLTAGE COEFFICIENT	PHASE SHIFT (DEGREES)
62	2 ms	2.6	2.4	0.92	– 9
125	1 ms	2.6	2.35	0.90	– 23
250	0.5 ms	2.6	2.1	0.81	– 36
625	0.2 ms	2.6	1.04	0.40	– 58.5
1.25k	0.1 ms	2.6	0.689	0.265	– 76.5

Table 3-2

39. Did the output waveform move to the left or to the right of the input waveform?

left

Is this a positive or negative phase shift?

positive

Is the phase shift for this RL circuit in the same direction as the phase shift for the RC circuit?

no

40. Phase shift ≈ **+27°**

41. Input voltage ≈ **2** Vp-p

Output voltage ≈ **1.52** Vp-p

Voltage coefficient ≈ **0.760**

44. Input ≈ **2** Vp-p

Output ≈ **1.414** Vp-p

Phase shift ≈ **+45°**

Voltage coefficient ≈ **0.707**

Is the voltage coefficient for a phase shift of 45 degrees approximately equal to the voltage coefficient for the RC circuit when its phase shift was 45 degrees?

yes

46. Phase shift ≈ **+10°**

Based upon your phase shift measurements in Steps 40, 44, and 46, which set of frequencies offer the *best* output voltage response — lower or higher frequencies?

higher

53. Is the output signal in-phase or out-of-phase with the input signal?

out-of-phase

55. Is this output in-phase or out-of-phase with the input signal?

in-phase

How do you account for this phase switching?

Placing the ground on the high side of the secondary makes it opposite to the grounded end of the primary, thus switching the phase of the transformer.

INSTRUCTOR NOTES

EXERCISE 4

Frequency Response Measurements

2. $f_{LOW} \approx$ **83** Hz

   (NOTE TO THE INSTRUCTOR: Tolerances of the parts used in the Trainer may not allow the Carrier Generator to go this low in the 1 kHz range. Any value less than 150 Hz is acceptable.)

3. $f_{HIGH} \approx$ **1090** Hz

4. $\approx$ **+1** dBm

9. Input (Channel 1) $\approx$ **2.5** Vp-p

10. Output (Channel 2) $\approx$ **1.16** Vp-p

11. Frequency (Output p-p MAX) $\approx$ **135** Hz

12. Input (Channel 1) $\approx$ **2.5** Vp-p

    Output (Channel 2) $\approx$ **2** Vp-p

13. Output $\approx$ **–5** dBm

14. (**NOTE TO THE INSTRUCTOR:** The values in Table 4-1 are approximations only. The values your students arrive at may differ, but there should be a decrease in the power level starting at 200 Hz, which occurs after the resonant frequency of the circuit.

Hz	100	200	300	400	500	600	700	800	900	1.0k	1.5k	2.0k	2.5k
dBm	– 5	– 5	– 5	– 6	– 6	– 7	– 8	– 9	– 9	– 10	–13	– 16	– 18

Table 4-1

INSTRUCTOR NOTES)

EXERCISE 5

Circuit Effects on Voice Signals

17. (**NOTE TO THE INSTRUCTOR:** There will be large variations in your student's values, due to the various acoustical resonances which occur. The values shown here are examples only.)

	FREQUENCY #1	FREQUENCY #2	FREQUENCY #3	FREQUENCY #4
FREQUENCY (Hz)	1738	1923	2042	1400
F GEN OUT (Vp-p)	2.1	2.1	2.1	2.1
TP122 (Vp-p)	4.5	3	2.4	2.2

Table 5-1

32. Frequency (Output = 300 mV p-p) ≈ **5000** Hz

39. Frequency (Output = 1.5 Vp-p) ≈ **6000** Hz

42. Frequency (Output = 1.5 Vp-p) ≈ **7200** Hz

INSTRUCTOR NOTES)

EXERCISE 6

Measuring Signal Waveforms and Harmonics

14. Center frequency ≈ **300** Hz

17. Low Freq. ≈ **298** Hz

    High Freq. ≈ **304** Hz

    BW ≈ **6** Hz

21. Low Freq. ≈ **285** Hz

    High Freq. ≈ **315** Hz

    BW ≈ **30** Hz

    Has the bandwidth increased?

    yes

30. Output ≈ **12** Vp-p

31. Output ≈ **+16** dBm

32. C GEN ≈ **45000** Hz

    Center Frequency ≈ **900** Hz

33. F GEN ≈ **300** Hz

    Based on your calculated center frequency and the FUNCTION GENERATOR frequency, which numbered harmonic have you just found?

    Third

37. Output ≈ **5** Vp-p

    Is the voltage level for this harmonic greater than or less than the voltage level for the third harmonic?

    less than

38. Output ≈ **+13** dBm

    Is the power level for this harmonic greater than or less than the power level measured in Step 31?

    less than

39. C GEN ≈ **75000** Hz

    Center Frequency ≈ **1500** Hz

    Based upon your calculated center frequency and the FUNCTION GENERATOR frequency (from Step 33), which numbered harmonic have you just found?

    Fifth

44. Output ≈ **2** Vp-p

    Is this voltage level greater than or less than the previous voltage levels at the other harmonics?

    less than

45. Output ≈ **+7** dBm

    Is the power level for this harmonic greater than or less than the power level for previous harmonics?

    less than

    What conclusion can you reach about the output levels for triangular harmonic frequencies?

    As the number of the harmonic increases, the output level decreases.

46. C GEN ≈ **105000** Hz

    Center Frequency ≈ **2100** Hz

    Based upon your calculated center frequency and the FUNCTION GENERATOR frequency (from Step 33), which numbered harmonic have you just found?

Seventh

47. (**NOTE TO THE INSTRUCTOR:** Table 6-1 contains the values found for Steps 30 through 46, as well as, the values for the ninth harmonic frequency of a triangular wave. The values shown are only examples — your students' values may differ, but the same general results should be obtained.)

	FREQUENCY (Hz)	HARMONIC NUMBER	VOLTAGE (Vpp)	POWER LEVEL (dBm)
FREQUENCY #1	900	3	12	+16
FREQUENCY #2	1500	5	5	+13
FREQUENCY #3	2100	7	2	+7
FREQUENCY #4	2700	9	1.5	–1

Table 6-1

51. (**NOTE TO THE INSTRUCTOR:** Table 6-2 contains the values found for the harmonic frequencies of a square wave. The values shown are only examples—your students' values may differ, but the same general results should be obtained.)

	FREQUENCY (Hz)	HARMONIC NUMBER	POWER LEVEL (dBm)
FREQUENCY #1	900	3	+16
FREQUENCY #2	1500	5	+16
FREQUENCY #3	2100	7	+16
FREQUENCY #4	2700	9	+13

Table 6-2

52. Which group of numbered harmonic frequencies are contained within a square wave — even, odd, or all?

odd

Describe the difference in power levels between the harmonic frequencies for triangular waves (Table 6-1) and the harmonic frequencies for square waves:

The harmonic frequencies for square waves have much higher power levels than triangular waves.

65. A. C GEN ≈ **60000** Hz

B. f_C ≈ **1200** Hz

C. Harmonic # = **Second**

D. Output ≈ **+8** dBm

E. (NOTE TO THE INSTRUCTOR: Table 6-3 contains the values found for the harmonic frequencies of a sawtooth wave. The values shown are only examples — your students' values may differ, but the same general results should be obtained.)

	FREQUENCY (Hz)	HARMONIC NUMBER	POWER LEVEL (dBm)
FREQUENCY #1	1200	2	+8
FREQUENCY #2	1800	3	+6
FREQUENCY #3	2400	4	+4
FREQUENCY #4	3000	5	+2
FREQUENCY #5	3600	6	0
FREQUENCY #6	4200	7	–1

Table 6-3

INSTRUCTOR NOTES

EXERCISE 7

Applications of Harmonic Signals

8. What has happened to the output signal? What is this type of signal called?

   The output signal has gone from a sine wave to a rectified sine wave, with the negative alternation completely removed.

9. What has happened to the output signal?

   The output signal has become clipped, removing the peaks of both alternations.

19. Center frequency ≈ **300** Hz

23. Output ≈ **+16** dBm

24. Are there any harmonic signals present?

   no

   Why?

   Because a sine wave has no harmonics.

28. What type of signal is applied to the filter?

   Negative half-wave signal.

   What happened to the output power and voltage levels when the input signal changed?

   Both the dBm and p-p voltage levels dropped.

   Output ≈ **+12** dBm

30. Output ≈ **+5** dBm

31. C GEN Frequency ≈ **30000** Hz

    f_C ≈ **600** Hz

    Which numbered harmonic have you found?

    Second

32. C GEN Frequency ≈ **60000** Hz

    f_C ≈ **1200** Hz

    Which numbered harmonic have you found?

    Fourth

33. Based upon your answers to Steps 31 and 32, describe the harmonic signature of a rectified sine wave:

    The harmonic signature (structure) of a rectified sine wave is even-numbered harmonics, whose power levels drop fairly quickly.

35. What type of signal is applied to the filter?

    A clipped sine wave.

    What happened to the output power and voltage levels when the input signal changed?

    Again, the dBm and p-p voltage levels dropped.

    Output ≈ **+8** dBm

36. Output ≈ **–5** dBm

37. C GEN Frequency ≈ **45000** Hz

    f_C ≈ **900** Hz

    Which numbered harmonic have you found?

    Third

39. C GEN Frequency ≈ **75000** Hz

    f_C ≈ **1500** Hz

    Which numbered harmonic have you found?

    Fifth

40. Based upon your answers to Steps 37 and 39, describe the harmonic signature of a clipped sine wave:

    The harmonic signature (structure) for a clipped sine wave is similar to a square wave — all odd-numbered harmonics. However, because the edges of the clipped waveform are not as sharp as for a square wave, the power levels of the harmonics are much lower than for a square wave.

53. $P_{fund} + P_{har} = \log_{-1}\frac{dBm}{10} =$ **31.62** mW

54. P_{har} ≈ **–6** dBm

55. P_{har} ≈ **0.251** mW

56. $T.H.D = \left(\frac{P_{har}}{P_{fund} + P_{har}}\right) \times 100 \approx$ **0.793** %

59. Output (filter tuned, rectified input) ≈ **+2** dBm

60. P_{har} ≈ **1.58** mW

61. $T.H.D = \left(\frac{P_{har}}{P_{fund} + P_{har}}\right) \times 100 \approx$ **4.996** %

    Is this percentage higher or lower than with a sine wave input?

    higher

    Explain why the T.H.D. value you just calculated differs from the T.H.D. value for a sine wave input:

    When harmonics within a signal have a high power level (as in a distorted sine wave, or a non-sine wave), the T.H.D. value is higher.

64. Output (filter tuned, clipped input) ≈ **+3** dBm

65. P_{har} ≈ **1.99** mW

66. T.H.D. ≈ **6.29** %

    Is this percentage higher or lower than with a sine wave input?

    higher

    Explain why the T.H.D. value you just calculated differs from the T.H.D. value for a sine wave input:

    This merely indicates that a clipped signal has greater harmonics than a rectified signal.

67. Now that you've calculated the T.H.D. for both the rectified and clipped inputs, can this value indicate the *type* of distortion, or merely its existence?

    Only its existence.

    Explain your answer:

    The T.H.D. value only shows whether distortion exists — it cannot define the type of distortion.

INSTRUCTOR NOTES

EXERCISE 8

Combining Signals — Intermodulation Distortion

3. Is the signal displayed by Channel 1 caused by a linear or a non-linear circuit?

linear

What is the term which identifies the signal from one source being added to a second signal within the same circuit?

Superposition

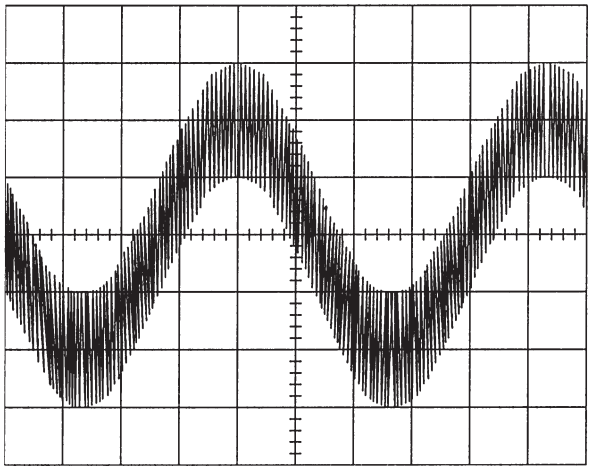

Figure 8-1

4. Describe the effect on the Channel 1 signal:

The level of the higher frequency signal changes.

5. Describe the effect on the Channel 1 signal:

The lower frequency signal changes its waveform in accordance to the setting of the FUNCTION switch.

6. Is this signal linear or non-linear?

non-linear

Is this signal balanced or unbalanced?

unbalanced

7. Is this signal linear or non-linear?

non-linear

Is this signal balanced or unbalanced?

balanced

12. Is the Channel 1 signal similar to the waveform shown in Figure 8-2A?

yes

13. R313 performs the same function as which control on the Trainer?

The OFFSET control.

By adjusting R313, what characteristic of the signal are you adjusting?

The balance.

29. Output ≈ **+14** dBm

30. C GEN ≈ **300** kHz

f_C ≈ **6000** Hz

Is the Tunable Filter tuned to the carrier input or the modulation input?

carrier

31. Output ≈ **+12** dBm

32. C GEN ≈ **260** kHz

    f_C ≈ **5200** Hz

    What is the term which defines the relationship between this frequency and the frequency you measured in Step 30?

    This is the *difference* frequency.

    Explain why the Tunable Filter causes a significant output level at this frequency:

    Because of intermodulation, the difference between the carrier and modulation frequencies will have a significant power level.

34. Output ≈ **+12** dBm

35. C GEN ≈ **340** kHz

    f_C ≈ **3800** Hz

    What is the term which defines the relationship between this frequency and the frequency you measured in Step 30?

    This is the *sum* frequency.

    Explain why the Tunable Filter causes a significant output level at this frequency:

    Because of intermodulation, the sum of the carrier and modulation frequencies will have a significant power level.

36. (**NOTE TO THE INSTRUCTOR:** The students' drawing in Figure 8-5 is based upon the power and frequency values from Steps 31 through 35. The example shown here is based upon the same values, and may differ from the student's drawing.)

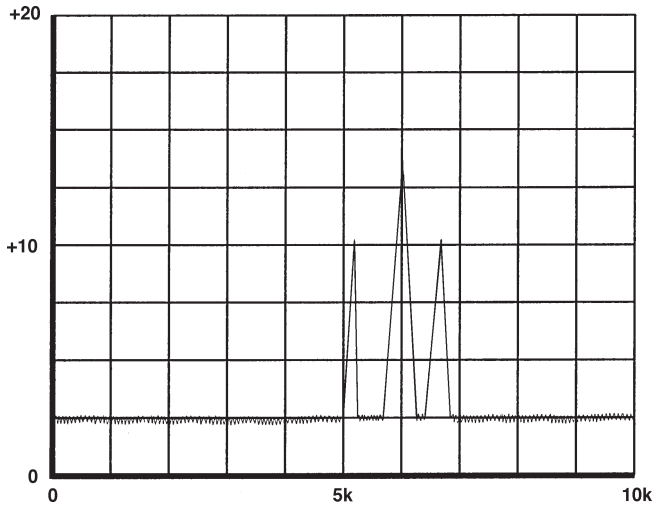

Figure 8-5

38. Will changing the FUNCTION GENERATOR frequency change any of the frequencies you found in Steps 29 through 35?

yes

If so, which frequencies will change, and why?

The sum and difference frequencies will change because the modulation frequency has changed.

39. Output ≈ **+14** dBm

40. $f_C \approx$ **6000** Hz

Did your adjustment of the FUNCTION GENERATOR change the frequency which causes the largest output of the Filter?

no

41. Output ≈ **+8** dBm

42. $f_C \approx$ **4940** Hz

Is this frequency different from the lower frequency you found in Step 32?

yes

Why?

Since the modulation frequency increased, the difference frequency will be lower.

44. Output ≈ **+8** dBm

45. $f_C \approx$ **7060** Hz

Based upon your frequencies found in Steps 42 and 45, what is the present frequency of the FUNCTION GENERATOR?

Approximately 1060 Hz.

How did you arrive at your answer?

It is the difference between the center frequency (found in Step 40) and the sum and difference frequencies (found in Steps 42 and 45).

46. (**NOTE TO THE INSTRUCTOR:** The students' drawing in Figure 8-6 is based upon the power and frequency values from Steps 39 through 45. The example shown here is based upon the same values, and may differ from the student's drawing.)

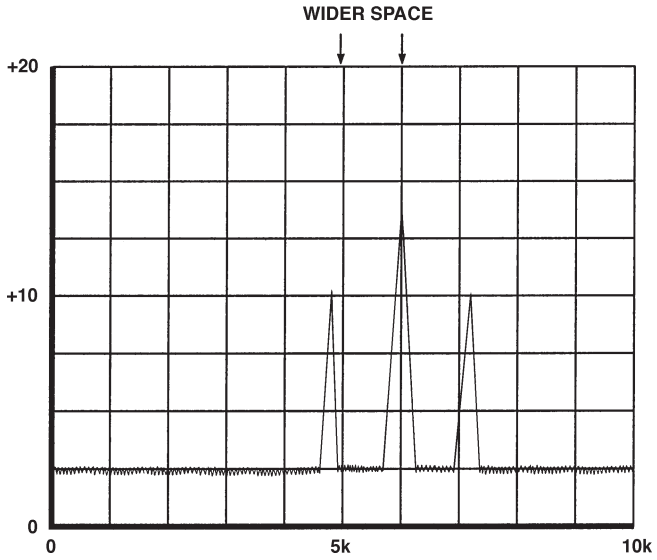

Figure 8-6

INSTRUCTOR NOTES

EXERCISE 9

Application of Intermodulation Signals

17. Output ≈ **+14** dBm

18. C GEN ≈ **295** kHz

    f_C ≈ **5900** Hz

    Square Wave Oscillator ≈ **5900** Hz

19. Does the Filter tune to the sum and difference frequencies of the Modulator?

    no

    Why?

    Since the FUNCTION GENERATOR level is fully CCW, there is no modulating frequency.

21. Does the level of the FUNCTION GENERATOR signal affect the level of the Filter output, when tuned to the carrier signal?

    no

23. Output (sum frequency) ≈ **+4** dBm

24. Sum frequency ≈ **7300** Hz

25. Does the FUNCTION GENERATOR level alter the output level when the Filter is tuned to the sum frequency?

    yes

27. Output (difference frequency) ≈ **+6** dBm

28. Difference frequency ≈ **4500** Hz

29. Does the FUNCTION GENERATOR level alter the output level when the Filter is tuned to the difference frequency?

yes

35. $f_C \approx$ **5580** Hz

36. $f_C \approx$ **6280** Hz

Based upon your values for Steps 35 and 36, what is the passband for the Tunable Filter?

Approximately 700 Hz.

37. Sum frequency (Step 24) ≈ **7300** Hz

Difference frequency (Step 28) ≈ **4500** Hz

Are the sum and difference frequencies within the passband of the Filter?

no

Are they easily located?

yes

41. Carrier frequency (Step 18) ≈ **5900** Hz

F GEN frequency (Step 39) = **200** Hz

Sum frequency ≈ **6100** Hz

Difference frequency ≈ **5700** Hz

Will these sum and difference frequencies be within the passband of the Filter?

yes

42. $f_C \approx$ **5500** Hz

43. $f_C \approx$ **6300** Hz

Based on your values for Steps 42 and 43, what is the passband for the Tunable Filter?

Approximately 800 Hz.

Did this have any effect of the passband of the Tunable Filter?

yes

44. Are you able to easily distinguish between the carrier frequency, and the sum and difference frequencies?

no

Imagine for a moment that these sum and difference frequencies were a part of an actual circuit. How would you modify the Filter so it would be able to separate the sum and difference frequencies from the passband?

Make the passband of the filter smaller.

49. Output (max unbalanced) ≈ **+14** dBm

51. Output (max balanced) ≈ **+3** dBm

Explain why balancing the Modulator output causes the output of the Filter to dramatically decrease:

The out-of-phase relationship in the balanced signal essentially cancels out the carrier frequency.

Will the balanced signal affect the existence of the sum and difference frequencies?

no

52. Difference frequency ≈ **+12** dBm

53. Sum frequency ≈ **+12** dBm

Are the sum and difference frequency power levels larger or smaller than the power level of the carrier frequency?

larger

INSTRUCTOR NOTES

EXERCISE 10

Transducer Specifications

12. F GEN frequency ≈ **1700** Hz

13. Describe in general terms the difference in tone quality between an undistorted (pure) sine wave and a distorted (clipped) sine wave.

    (**NOTE TO THE INSTRUCTOR:** This description will vary from student to student, but words such as "tinny" or "shrill-like" should be used.)

14. Output ≈ **22** Vp-p

    Output ≈ **100.8** mW

    Output (dBm) = 10 $\log_{10}$ × Output (mW) ≈ **20** dBm

15. Output (with load) ≈ **5** Vp-p

a. What is the mathematical relationship between the unloaded and loaded voltages?

   The loaded voltage is approximately one-half of the unloaded voltage.

b. Based upon your answer above, what must have happened to the total output impedance when you added the 620-ohm load?

   The total output impedance must have dropped by one-half.

c. Since you know the load resistance and the change in total output impedance, what was the output impedance when no load was attached?

   Approximately 620 ohms.

20. Output (–20 dB) ≈ **2** Vp-p

    Is the signal free from ripple at this point?

no

21. Output (–40 dB) ≈ **0.2** Vp-p

22. Output (–80 dB) ≈ **0.002** Vp-p

    With the signal at this level, what dominates the Channel 1 display?

    The ripple.

23. Output (J102 open) ≈ **0.14** Vp-p

    With J102 open, you have removed both the signal *and* the ripple from the scope display. Therefore, what must be causing the Channel 1 waveform?

    The noise in the circuit.

    You have either measured or calculated the signal voltage level for each 20 dB drop. At approximately what dB point would the signal voltage level have equaled the waveform presently displayed on your scope?

    Approximately –59 dBm.

    What does this dB level represent?

    The point at which the signal disappears into the noise.

24. Dynamic range ≈ **79** dB

32. Voltage to Speaker ≈ **0.3** Vp-p

33. A. Output = **1.4** Vrms

    B. Voltage ratio ≈ **1.4**

    C. Speaker Output (4 Vp-p) ≈ **87** dB_{SPL}

    D. Voltage to Speaker ≈ **0.106** Vrms

    E. Power to Speaker ≈ **1.4** mW

    F. Power ratio ≈ **714**

    G. Speaker Efficiency (n) ≈ **112** dB_{SPL}/W

35.

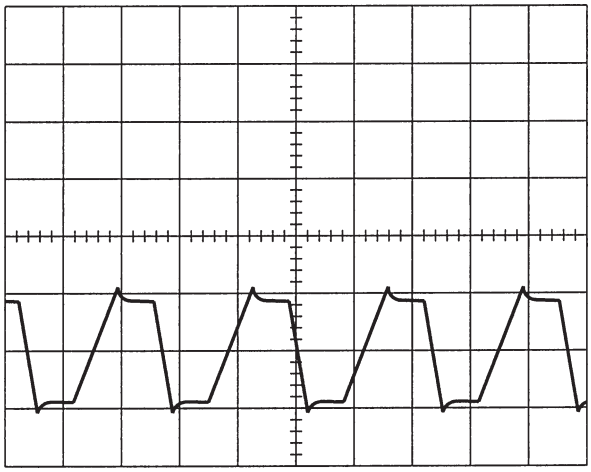

Figure 10-7

37. (**NOTE TO THE INSTRUCTOR:** The values in Figure 10-8 are examples only—the values measured by your students may differ.)

FREQUENCY (Hz)	POWER LEVEL (dBm)	DISTORTION? (Y/N)
200	N/A	N
400	+4	N
600	–4	Y
800	–4	Y
1000	–5	Y
1200	–1	Y
1400	+1	N
1600	+4	N
1800	+2	N
2000	+2	N

Figure 10-8

Which frequency had maximum distortion?

(This answer could be anywhere from 600 Hz to 1200 Hz.)

38.

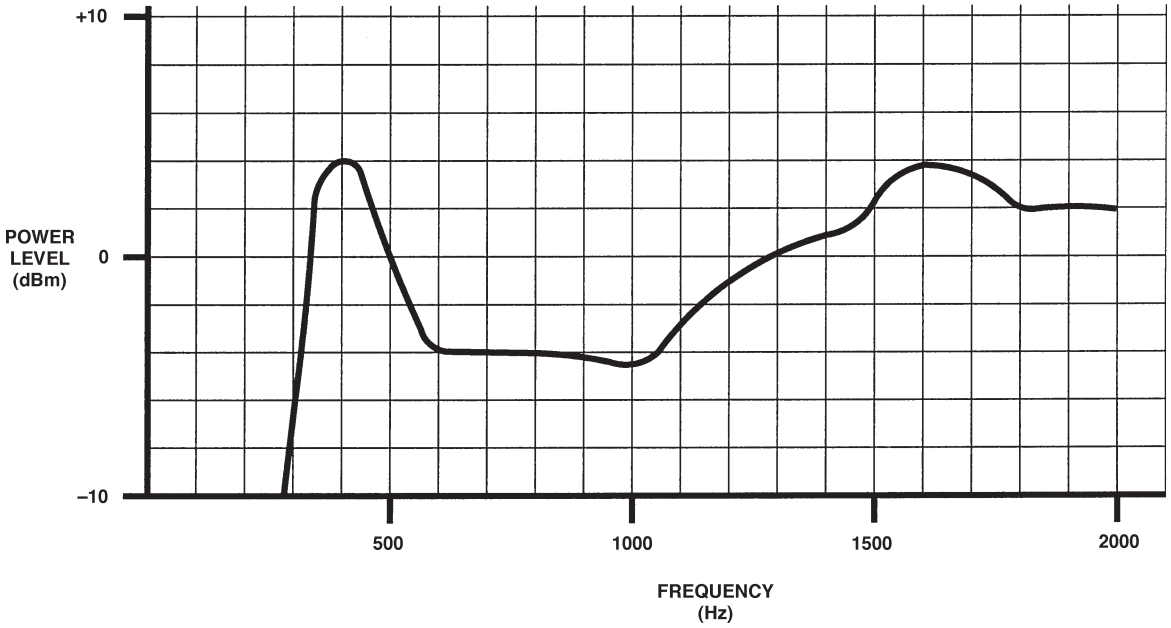

Figure 10-9

41. Acoustical resonances occur at the following frequencies:

Approximately 400 and 1600 Hz.

44. A. TP204 (Amp. Out) ≈ **2** Vp-p

B. E_{LS1} ≈ **0.58** Vp-p

C. E_{R211} ≈ **1.42** Vp-p

D. Voltage ratio ≈ **0.408**

E. Speaker Z ≈ **19 Ω**

45.

Frequency = 300 Hz	**Frequency = 500 Hz**
TP204 ≈ **1.6** Vp-p	TP204 ≈ **1.75** Vp-p
E_{LS1} ≈ **0.28** Vp-p	E_{LS1} ≈ **0.3** Vp-p
E_{R211} ≈ **1.32** Vp-p	E_{R211} ≈ **1.45** Vp-p
Voltage ratio ≈ **0.212**	Voltage ratio ≈ **0.206**
Speaker Z ≈ **9 Ω**	Speaker Z ≈ **9.7 Ω**

52. (**NOTE TO THE INSTRUCTOR:** The values in Figure 10-14 are examples only — the values measured by your students may differ.)

FREQUENCY (Hz)	POWER LEVEL (dBm)	DISTORTION? (Y/N)
200	N/A	N
400	–1	N
600	+1	N
800	+4	N
1000	+7	N
1200	+5	N
1400	+8	N
1600	+5	N
1800	+8	N
2000	+7	N

Figure 10-14

53.

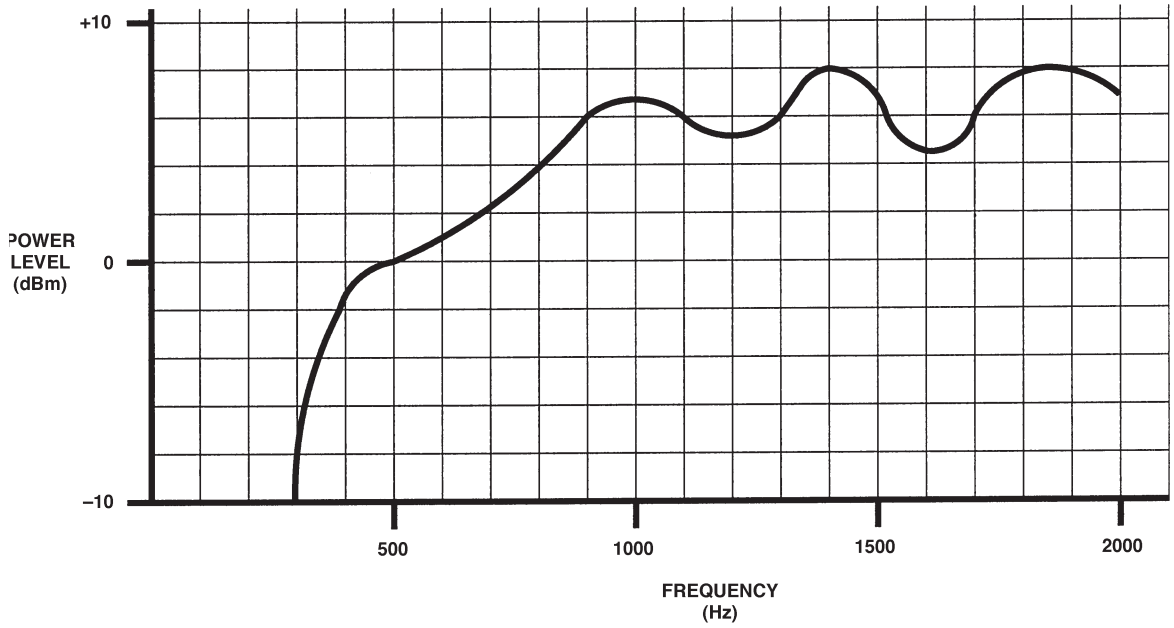

Figure 10-15

54. A. Based upon *output power levels only*, can you say that the Telset #1 receiver is **better** than the Backpack Speaker?

no

Why?

Since the input level was higher to begin with, the output levels will also be higher. This does not mean that the output transducer is better.

B. Based upon the distortion characteristics in the two frequency response tests, which output transducer is better — the Telset #1 receiver or the Backpack Speaker?

The Telset #1 receiver.

How did you arrive at your answer?

There is far less distortion in the Telset #1 test.

56. Output ≈ **4.2** Vp-p

57. Telset Receiver Efficiency (n) ≈ **114.5** dB_{SPL}/W

Compare the efficiency ratings of the Backpack Speaker and the Telset #1 receiver. Based upon these values, which is a more efficient output transducer?

The Backpack Speaker.

What do you think could account for this?

The telset receiver is designed to reproduce sound for the human ear, which is placed very close to the receiver. The Backpack Speaker is not designed for close proximity work, so it should be able to produce a higher sound level per watt of input.

INSTRUCTOR NOTES

SELF-TEST REVIEW ANSWERS

Section 1

CHAPTER 1

1. D — The difference between AC as applied to power and AC as applied to signals is **the frequency**.

2. C — The difference between acoustic and electric information signals is **their audibility**.

3. The electrical component that converts signals from acoustic-to-electric, or electric-to-acoustic form is called a **transducer**.

4. The technical field that uses electrical signals to communicate information is called **telecommunications**.

5. The shortened form of the term "electromagnetic radiation" is **radio.** Radiation.

6. C — FM stereo broadcast frequencies are within the **VHF** band.

7. A — AM broadcast frequencies are within the **MF** band.

8. A — The **EHF** band has the shortest wavelengths of the electromagnetic spectrum.

9. The approximate velocity for acoustic waves in air at sea level is **1100 ft/s** and **330 m/s**. This is the speed of sound.

10. The approximate velocity of electrical waves in wires is **984 $\times$ 10^6 ft/s** and **300 $\times$ 10^6 m/s**.

11. The dimensions for the phase of half of a signal wavelength are **0.5 (one-half)** cycle, **180** degrees, and **π (pi)** rads.

12. The unit measure of frequency is **time**.

13. The unit measure of wavelength is **distance**.

14. Phase is measured in **degrees** or **parts of a wavelength**.

15. B — Electrical noise will interfere most with a signal **when it is near the signal's power level.**

16. The wavelength of a 3 kHz electrical signal is:

$$\text{wavelength} = \frac{\text{velocity}}{\text{frequency}} = \frac{984 \times 10^6}{3 \times 10^3} = \mathbf{3.28 \times 10^5 ft}$$

$$\text{wavelength} = \frac{\text{velocity}}{\text{frequency}} = \frac{300 \times 10^6}{3 \times 10^3} = \mathbf{1 \times 10^5 m}$$

17. $\text{Frequency} = \frac{1}{\text{time}} = \frac{1}{5 \times 10^{-3}} = \mathbf{2000\ Hz}$

18. The time period-per-cycle of a 20 kHz signal is:

$$\text{time} = \frac{1}{\text{frequency}} = \frac{1}{20 \times 10^3} = \mathbf{50\ \mu s}$$

19. The time period for one radian of a resonant electrical network with 0.5 mH inductance and 2 µF capacitance is:

$$\text{time} = \sqrt{LC}\ \sqrt{(5 \times 10^{-4}) \times (2 \times 10^{-6})} = \mathbf{31.6\ \mu s}$$

20. The time period-per-cycle of a resonant electrical network with 1 mH inductance and 2 µF capacitance is:

$$\text{time} = 2\pi\sqrt{LC} = 2\pi\sqrt{(1 \times 10^{-3}) \times (2 \times 10^{-6})} = \mathbf{281\ \mu s}$$

21. The frequency of a resonant electrical network with 5 mH inductance, and 1 µF capacitance is:

$$\text{time} = \frac{1}{2\pi\sqrt{LC}} = \frac{1}{2\pi\sqrt{(5 \times 10^{-3}) \times (1 \times 10^{-6})}} = \mathbf{2252\ Hz}$$

CHAPTER 2

1. B — When multiplying rms volts × rms amps, with signal voltage and current in phase, the resulting product is said to be **true power**.

2. **1** (components) A. power consumption
   2 (signals) B. power level
   1 (components) C. power gain
   1 (components) D. power loss

3. A — The most practical data for calculating signal power are the signal's **voltage** and the circuit's **resistance**.

4. D — The most commonly used impedance for audio signal transmission circuits is **600 ohms**.

5. C — In car radios, power boosters increase signal power to the speakers by **increasing the power supply voltage above the 12 V** provided by the car battery.

6. B — The true power within any circuit that is not entirely resistive is less than the amps × volts within that circuit. True power is mathematically related to the proportion of reactance to resistance within the circuit, and calculated as:

   True power = amps × volts × $\cos \angle \theta$

   Since the reactance and resistance are equal, $\angle \theta$ is 45° . The cosine of 45° is about 70% (0.707); so the true power is about **30%** less than the apparent power in the circuit.

7. A — A large portion of true power is dissipated in the **resistive network**.

8. D — Power gains and losses are ratios of two signal power levels. As such, they are specified **in factors**, not watts or volts.

9. The power dissipated in an 8 ohm loudspeaker with an 8 volt p-p signal across its terminals is:

$$P = \frac{Vp\text{-}p}{8\,Z} = \frac{8^2}{8 \times 8} = \mathbf{1\ W}$$

10. The signal power when 8 volts rms is measured across a 600 ohm transmission circuit is:

$$P = \frac{Vrms^2}{Z} = \frac{8^2}{600} = \mathbf{107\ mW}$$

11. The signal voltage required for a reference power level of 1 milliwatt on a 600 ohm audio circuit is:

$$Vp\text{–}p = \sqrt{P\ 8R} = \sqrt{0.001 \times 8 \times 600} = \mathbf{219\ Vp\text{–}p}$$

12. The signal voltage required to drive an 8 ohm loudspeaker at a power level of 50 watts is:

$$Vrms = \sqrt{PR} = \sqrt{50 \times 8} = \mathbf{20\ Vrms}$$

13. The power factor of a network with 300 ohms resistance and 200 ohms reactance is:

$$\text{Power Factor} = \cos\left[\tan^{-1}(200 \div 300)\right] = \mathbf{0.832 = 83.2\%}$$

14. With 8 volts rms applied to an RL network of 100 ohms resistance and 100 ohms reactance, the true power dissipated by the network is:

$$\text{Phase angle} = \left[\tan^{-1}(100 \div 100)\right] = \mathbf{45°}$$

$$P = \frac{8^2}{100} \cos\ 45° = \mathbf{45\ mW}$$

CHAPTER 3

1. D — The intensity of all physical phenomena, including light, sound, and electrical signals, vary in logarithmic units of intensity. **Monetary exchange** is not logarithmically scaled.

2. A — A **3 dB** difference in sound level is just perceptible.

3. B — It takes an increase of **10 dB** to make a signal significantly louder.

4. D — **60 dB** represents a power ratio of one million to one.

5. D — An amplifier with 13 dB gain, and 2 mW applied to its input, has a gain factor of 20. The power at its output is 20 × 2 mW, or **40 mW**.

6. C — An attenuator that reduces 2 mW at its input to 0.1 mW at its output has a loss factor of 20. The logarithmic value of this factor is **13 dB**.

7. D — A gain of 19 dB is equivalent to a power ratio of **80:1**.

8. B — A power ratio of 40:1 is equivalent to a gain of **16 dB**.

9. C — A voltage ratio of 100:1 is equivalent to a power gain of **40 dB** with matched impedances.

10. C — When a 2 mW signal is passed through a 20 dB amplifier it is amplified by a power ratio of 100, to a level of 200 mW. The 200 mW signal is reduced to half that value as it passes through the 3 dB attenuator. The output is **100 mW**.

11. C — **3 dB** is the referenced power level of a 2 mW signal.

12. B — 13 dB is a power ratio of 20. 13 dB above a 1 mW reference is 20 × 1 mW, or **20 mW**.

13. C — A 6 dBm signal has a power level of 4 mW. When amplified by 10 dB, the 4 mW level is increased by a factor of 10, to **40 mW**.

14. A — 1 pW is 9 decimal places less than 1 mW. 100 pW is 7 decimal places less than 1 mW. Seven decimal places are equal to a power ratio of 70 dB. 100 pW is a power level of **–70 dBm**.

15. C — A power level of –50 dBm is 5 decimal places less than 0 dBm, or 1mW. Six decimal places less than 1 mW is 1 nW. Five decimal places less than 1 mW is **10 nW**.

16. A signal increase from 2 mW to 500 mW is a gain of **24 dB**.

17. A signal decrease from 2 mW to 0.5 fW is a loss of **–36 dB**.

18. The power gain of an amplifier with an input of 5 mV and an output of 1 V is **46 dB**.

19. The power gain of an amplifier that increases a signal voltage by a factor of 80 is **38 dB**.

20. In an impedance matched amplifier, a voltage ratio of **20:1** will result in a 26 dB gain.

21. The power level of a 24 dBm signal is **251 mW**.

22. The power level of a –24 dBm signal is **3.98 μW**.

23. The referenced power of a 50 pW signal is **–73 dBm**.

24. The referenced power of a 50 W signal is **47 dBm**.

25. A receiver with a sensitivity of 20 dBf has a power level of **100 fW**.

CHAPTER 4

1. A — Sound pressure levels are measured in **dB**.

2. A — Signal gain and loss are measured in **dB**.

3. D — The human ear is most sensitive to soft sound signals at frequencies around **4000 Hz**.

4. B — The spectrum analyzer doesn't indicate **time** quantities.

5. D — **Network analyzers** provide a continuous graph of the frequency response of electronic circuits.

6. C — **Spectrum analyzers** provides a graphical display of individual signal power levels over a range of frequencies.

7. C — With a 10 mW signal applied to the input of a filter with 20 dB of rejection at that frequency, the output power will be **0.1 mW**.

8. D — With a corner frequency of 2 kHz, 8 kHz is 2 octaves above that frequency, thereby attenuating the signal 2 × 6 dB, or 12 dB. The signal is attenuated by 3 dB at the corner frequency; so the total attenuation is 12 dB + 3 dB, or **15 dB**.

9. B — A transmission circuit with a "first pole" (corner frequency) at 2 kHz will reduce the level of higher frequency signals by 20 dB-per-decade. 20 kHz is one decade above (10×) 2 kHz. The signal is attenuated by 3 dB at the corner frequency; so the total attenuation is 20 dB + 3 dB, or **23 dB**.

10. B — The reactance at the corner frequency is equal to the resistance of an RC or RL network. The equivalent resistance of a 600 ohm transmission circuit is 600 ohms; thus, the reactance is also **600 ohms**.

11. B — The phase shift at the corner frequency, where reactance and resistance are equal, is **45°**.

12. C — With a 40 dB S/N, and a noise power of 200 nW, the signal is 40 dB above 200 nW, or **2 mW**.

13. A — With a sensitivity of 22 dBf, and a S/N of 24 dB, the noise power is 24 dB below 22 dBf, or **–2 dBf**.

14. B — Maximum power transfer occurs when source, cable, and load impedances are matched. Computer equipment is most often standardized at **75 ohms**.

15. A — A **0.5 dB** loss will occur when a 300 ohm transmission line is used in a circuit designed for 600 ohm lines.

16. The attenuation of a transmission circuit with a 2 kHz first pole frequency, and a signal frequency of 40 kHz is **29 dB**.

17. With a 2 mW signal level and a 100 pW noise level, the S/N is **73 dB**.

18. The signal level required to be 30 dB above a noise level of 100 μW is best determined by reckoning:

    30 dB = 10^3 = 1000 times the noise level.
    30 dB above 100 μW is **100 nW**; or **0.1 μW**.

19. A — **4 dB** mismatch loss will occur if a 75 ohm cable is used in a 600 ohm transmission circuit.

Section 2

CHAPTER 5

1. D — The voltage rises to a positive maximum at **90°** in a sine waveform.

2. B — When a sine wave of one frequency is combined with another three times its frequency, the maximum positive voltage in the waveform occurs where both signals are at positive polarity. This occurs at **30°** of the lower frequency signal, which is 90° of the higher frequency signal.

3. C — A signal with a 400 Hz fundamental frequency has a third harmonic at three times that frequency, or **1200 Hz**.

4. A — "Complex" waveforms have harmonic frequencies included within their structure. Only pure **sine** waves do not have harmonics.

5. C — Of the waveforms listed, the **square** wave has predominantly odd-numbered harmonics at higher power levels.

6. A — Sawtooth waveforms contain all (even and odd) harmonic numbers; so the fourth harmonic above the fundamental is the fifth harmonic. The spectral width extends from 400 Hz (the fundamental) to 2000 Hz (5×400 Hz), which is **1600 Hz** wide.

7. With a fourth harmonic at 2000 Hz, the fundamental frequency is 1/4th of 2000 Hz, or **500 Hz**.

8. D — Both the oscilloscope and spectrum analyzer can measure frequency and power. The scope can also display the waveform shape. However, only the spectrum analyzer provides accurate indications of **harmonic distortion** by displaying both the power levels and numbers of the harmonic frequencies.

9. A — Any waveform applied to an LC network that is resonant to its fundamental frequency will be converted to a **sine** waveform due to the action of the circulating current within the LC network.

10. A — The harmonic with the highest power level in a rectified sine waveform is that closest to the fundamental, the **second** harmonic.

11. A sine wave with a peak voltage level of 5 volts, has instantaneous voltages in proportion to the sine of the phase angle of the waveform.

    A. 0° = 5 sin 0° = **0 volts**
    B. 30° = 5 sin 30° = **2.5 volts**
    C. 45° = 5 sin 45° = **3.5 volts**
    D. 60° = 5 sin 60° = **4.3 volts**
    E. 90° = 5 sin 90° = **5 volts**

12. Two frequencies are combined with the higher being 3× the lower frequency. The lower frequency has a peak voltage of 5, and the higher frequency has a peak voltage of 2. The instantaneous voltages of both waveforms are combined at the increments indicated for the lower frequency waveform, as follows:

    A. At 0°: 5 sin 0° = 0, + (2 sin 0° = 0) = **0 volts**
    B. At 30°: 5 sin 30° = 2.5, + (2 sin 90° = 2) = **4.5 volts**
    C. At 45°: 5 sin 45° = 3.5, + (2 sin 45° = 1.4) = **4.9 volts**
    D. At 60°: 5 sin 60° = 4.3, + (2 sin 0° = 0) = **4.3 volts**
    E. At 90°: 5 sin 90° = 5, + (2 sin -90° = -2) = **3 volts**

13. Referring to Figure 5-3 in the Textbook, the power ratio (in dB below the fundamental) for the third harmonic of each of the following waveforms, by examination, is:

    A. Square wave: **–10 dB**
    B. Pulse wave: **–3 dB**
    C. Triangle wave: **–15 dB**
    D. Sawtooth wave: **–12 dB**
    E. Rectified sine wave: **–40 dB**

CHAPTER 6

1. D — The power level of harmonic signals 50 dB below the level of the fundamental are usually considered insignificant. The power level **0.01 mW** is 50 dB below (10^{-5}) the level of the 1 watt fundamental.

2. C — In reference to Figure 5-3, the rectified sine wave signal has only even-numbered harmonics. The second harmonic signal above the fundamental is the **fourth** harmonic.

3. D — In reference to Figure 5-3, the square wave signal has only odd-numbered harmonics. The second harmonic signal above the fundamental is the **fifth** harmonic.

4. With the highest harmonic number of 5, the frequency distribution of a 400 Hz signal (fundamental) extends from **400 Hz through 2000 Hz** (5 × 400).

5. B — With the highest harmonic number of 7, the frequency distribution of a 400 Hz signal (fundamental) extends from 400 to 2800 Hz (7 × 400). The bandwidth is **2400 Hz**.

6. C — With a circuit bandpass from 350 Hz through 2050 Hz, the highest harmonic number of a 400 Hz signal that will pass through the circuit is the **fifth**. 5 × 400 Hz = 2000 Hz.

7. B — With a total harmonic power of 0.1 watt within the spectrum of a 1-watt fundamental signal, the T.H.D. (total harmonic distortion) is calculated as:

$$\text{T.H.D.} = \frac{P_{har}}{P_{har} + P_{fund}} \times 100 = \frac{0.1}{1 + 0.1} \times 100 = 0.09 \times 100 = \mathbf{9\%}$$

8. Distorting a signal in a way that reduces its harmonics requires that the bandpass of a circuit be reduced. The process that reduces circuit bandpass is called **filtering**.

9. Referring to Textbook Figure 6-1B, there are four harmonic signals that are less than 30 dB below the level of the fundamental; the second, third, fourth, and fifth (**four** harmonic signals).

10. Referring to Textbook Figure 6-1B, the bandpass required to pass all frequencies from the fundamental through the fifth harmonic, with a fundamental frequency of 400 Hz, must extend from 400 Hz through 5 × 400 Hz = 2000 Hz. The bandpass is **1600 Hz**.

11. With a harmonic signal indicated at 26 dB below a 1-watt signal, its power level is calculated as:

$$P_{har} = P_{fund}\left(\log_{-1}\frac{-\text{dB}}{10}\right) = 1\text{W}\left(\log_{-1}\frac{-26}{10}\right) = \mathbf{2.5\ mW}$$

12. Using the same procedures as in Problem 24, the power levels of the second, third, fourth, and fifth harmonic signals identified in Figure 6-1B of the Textbook are calculated using a 2-watt power level for the fundamental frequency, as:

    1. **632 mW**
    2. **63 mW**
    3. **200 mW**
    4. **7.9 mW**

    T.H.D. = **31.1%**

13. With a total harmonic power of 370 mW, and a 800 mW fundamental signal, the T.H.D. is **31.6%**.

14. With all harmonics but the second filtered, leaving only 200 mW of harmonic power, the reduced harmonic distortion is **20%**

.

CHAPTER 7

1. C — With a passband from 500 to 2000 Hz, the bandpass of this filter is 2000 – 500 = **1500 Hz**.

2. D — With a bandpass of 2 octaves, and a lower frequency limit of 500 Hz, the higher frequency limit is 2 octaves above 500 Hz. With each octave 2× its lower frequency limit, the high frequency limit is 2 × 2 × 500 Hz = **2000 Hz**.

3. B — With second and third harmonics included within the bandwidth of a 500 Hz fundamental frequency, the total bandwidth extends from 500 Hz through 3 × 500 Hz = 1500 Hz, a bandwidth of **1000 Hz**.

4. C — With the power at two points on the skirt (rejection slope) of a filter, 100 mW at 500 Hz, and 10 mW at 1000 Hz, and with 1000 Hz at 1 octave (twice the frequency) above the 500 Hz point, the power slope is 10:1, or **10 dB-per-octave**.

5. C — Filters that pass lower frequencies, but reject higher frequencies are called **low-pass** filters.

6. A — Filters that pass a specific range of frequencies, but reject frequencies above and below this range are called **bandpass** filters.

7. C — The harmonics in a signal's waveform occupy frequencies higher than the fundamental. A **low-pass** filter tuned slightly above the fundamental frequency will reject all harmonics.

8. A — A parallel LC network placed across a circuit will have high impedance and low loss at resonance, but low impedance and high loss above and below the resonant frequency, the properties of a **bandpass** filter.

9. B — A lower value of resistance in an LC resonant network will result in an **increased Q and higher selectivity**.

10. C — **Reduced Q and lower selectivity** results in a wider (larger) bandwidth.

10. A — An equalizing amplifier with an oppositely-sloping frequency response, called "**tilt**," is used to compensate for the higher frequency power losses caused by long cable runs.

12. The bandpass, in octaves, of a bandpass filter with frequency limits of 400 and 3400 Hz is calculated as:

$$PB = \frac{\log\left(f_{hi} \div f_{low}\right)}{0.3} = \frac{\log\left(3400 \div 400\right)}{0.3} = \mathbf{3.1\ octaves}$$

13. The % bandpass of a narrow-band filter (of less than an octave), with frequency limits of 56 and 80 kHz, is calculated using the same formula as the last problem; then, multiply by 100 to give a percentage of **51.6%**.

14. With series resonance at 3000 Hz, an LC network has 1000 ohms reactance and 20 ohms resistance. (At resonance both inductive and capacitive reactances are equal.) The Q, bandwidth, and passband range of frequencies of the filter are calculated as:

$$\mathbf{Q} = \frac{X}{R} = \frac{1000}{20} = \mathbf{50;\ BW} = \frac{F_o}{Q} = \frac{3000}{50} = \mathbf{60\ Hz}$$

$$f_{hi} = f_o\left(1 + \frac{1}{2Q}\right) = 3000\left(1 + \frac{1}{2 \times 50}\right) = 3030\ \text{Hz}$$

$$f_{low} - f_o\left(1 - \frac{1}{2Q}\right) = 3000\left(1 - \frac{1}{2 \times 50}\right) = 2970\ \text{Hz}$$

Passband range = 2970 Hz to 3030 Hz.

15. The skirt response of a filter produces rejections of 10 dB at 500 Hz and 20 dB at 800 Hz. The slope (selectivity) of the skirt is calculated in dB/octave, by first calculating the octave range, 0.68 octaves. With a loss of 10 dB (from 10 dB to 20 dB) in 0.68 octaves, the slope/octave is now calculated, by ratio and proportion, as:

$$\text{Slope} = \frac{dB_2 - dB_1}{BP} = \frac{20 - 10}{0.68} = \textbf{14.7 dB/octaves}$$

CHAPTER 8

1. A — Wideband circuits allow all significant level signals and their harmonics to pass, assuring the **accuracy** of the signal waveforms applied to the circuits.

2. B — A narrow band filter tuned to the fundamental frequency of a square wave signal will remove all harmonics from that signal's waveform. Signals with no harmonics have **sine** waveforms.

3. By heavy filtering, any harmonics that may occur in the waveforms of radio transmitters will be removed, **assuring that only the assigned frequency will be transmitted**.

4. The distortion that results in increased harmonics as a signal passes through a non-linear circuit is simply called **harmonic distortion**.

5. A — Many waveforms may appear at the output after passing a sine wave signal through a non-linear circuit, but a **perfect sine wave** will not appear when the signal is distorted.

6. C — A **square** waveform has primarily odd-numbered harmonics, with higher harmonic power in its third harmonic, and very low power second and fourth harmonics.

7. With a circuit low-pass filtered at 5 kHz, and the fifth harmonic required to retain accuracy in a signal waveform, a **1 kHz** signal will produce a 5 kHz fifth harmonic frequency.

8. All harmonics in a comb generator are spaced at the frequency of the fundamental, **10 kHz** for this example.

9. A marker generator spaces frequencies 5 kHz apart between 100 and 125 kHz. The fundamental frequency of the generator is thus **5 kHz**.

10. The advantages of frequency multiplication are that **only a single generator (oscillator) is required** to produce mixer frequencies for the multiplexer, and all **signals will track the accuracy and stability of the single generator**.

11. A — The ideal waveform for a frequency doubler is one that has a higher power second harmonic signal, like the **rectified sine** waveform.

12. A — The ideal waveform for a frequency tripler is one that has a higher power third harmonic signal, like the **square waveform**.

13. A — The output filtering and input frequency limits of a ×7 frequency multiplier require identification of the adjacent **sixth and eighth** harmonics. Higher fundamental frequencies space these harmonics further apart, making the necessary filtering that rejects them simpler to design.

14. B — The harmonics created by the pulse waveform of Figure 5-3B and Table 8-1 indicate higher harmonic power at only 1 dB below the level of the fundamental for the second and fourth harmonic. Of the choices, the most efficient is the fourth harmonic, or ×**4**.

15. Referring to Table 8-1 and Figure 5-3D, the harmonic structure of the sawtooth waveform shows a "slope," with a reduction in each harmonic's power level. Table 8-1 accurately indicates these losses. An equalizing amplifier will require a "tilt" of opposite character to raise the power level of each harmonic signal to equal the fundamental. The response of the amplifier is summarized in Figure 8-19.

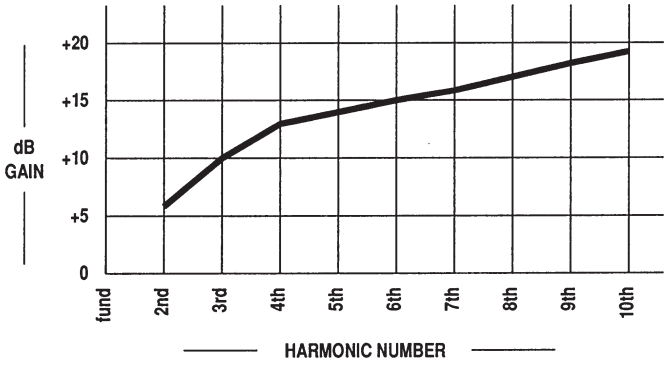

Figure 8-19

16. A ×5 frequency multiplier can be most efficiently designed using a circuit that clips a signal waveform to appear like a square wave with suppressed fourth and sixth harmonics. A practical filter can then be used to reject the third and seventh harmonics from the multiplier's output while maintaining a bandpass that allows an input signal variation from 9 kHz through 11 kHz.

    The third harmonic of an 11 kHz input signal applied to the ×5 multiplier is 33 kHz. The seventh harmonic of a 9 kHz input signal applied to the ×5 multiplier is 63 kHz. The ×5 frequency range of the 9 kHz to 11 kHz signals extends from 45 kHz through 55 kHz.

    A filter is required that **passes frequencies from 45 through 55 kHz, but rejects 33 kHz and 63 kHz signals**.

17. Table 8-1 shows the efficiency of a rectified sine wave doubler. The response indicates a second harmonic signal level at –3 dB in reference to the fundamental frequency applied to the doubler's input. With a 3 dB loss, the conversion efficiency is 50%, or **–3 dB**.

Section 3

CHAPTER 9

1. C — When two input signals are combined in a linear circuit, **only the two applied signals** will appear in the waveform at the circuit's output. Linear circuits don't create harmonics or any other frequencies.

2. C — If two sources apply +4 and +6 volts across a resistor individually, then together they will apply **+10 volts** across the resistor.

3. The name of the technique used to solve electrical network problems, such as described in Question 2, is "**superposition**."

4. B — The two signals that are applied across the T-network result in a load voltage that can be determined using superposition techniques. However, the signals are changing in phase, and they may be specified in rms, peak, or p-p units. In this example, the request is for voltages in peak units. At the instant when the lower-frequency signal is crossing zero, the only signal voltage across the network is from the higher-frequency source. Thus, the network will operate as though it was configured with S1 switched to shunt G1, placing R1 in parallel with R3. The load voltage across R3 will equal that portion of the 1 volt p-p of G2 not dropped across series resistance R2:

$$E_{R3} = E_{G2} [(R1 \,||\, R3) \div (R2 + R1 \,||\, R3)]$$

5. Refer to Figure 9-4.

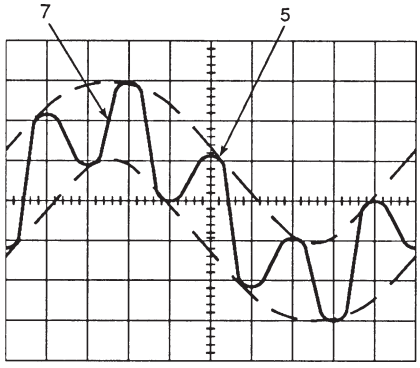

Figure 3-4

6. C — This problem is similar to that of Problem 4, but the two generators are reversed. At the instant when the higher-frequency signal is crossing zero, the only signal voltage across the resistor is from the lower-frequency source. Thus, the network will operate as though it was configured with S2 switched to shunt G2, placing R2 in parallel with R3. The load voltage across R3 will equal that portion of the 3 volts p-p of G1 dropped across the series resistance of R1:

$$E_{R3} = E_{G1} [(R2 \,||\, R3) \div (R1 + R2 \,||\, R3)]$$

$$= 3 [500 \div (1\text{ k} + 500)] = 1.0 \text{ Vp-p} \text{ or } \mathbf{0.5\ V\ peak}$$

7. Refer to Figure 9-4.

8. A — **Linear** circuits don't distort signal waveforms, and they don't create harmonics or any other frequencies.

9. The techniques of calculation are similar to those described for Problems 4 and 6:

   A. $E_{R3} = \mathbf{0.67\ Vp\text{-}p}$

   B. $E_{R3}' = \mathbf{0.33\ Vp\text{-}p}$

   C. $E_{R3} = (0.67 \text{ Vp-p} \div 2) + (0.33 \text{ Vp-p} \div 2) = \mathbf{0.5\ Vpeak}$

   D. $E_{R3} = (+0.17) + (-0.33) = \mathbf{0.5\ Vpeak}$

10. The values of the two signals in the waveform can be calculated as follows; using E1 and f1 for the voltage and frequency (time C) of the lower-frequency signal, and E2 and f2 for the voltage and frequency (time B) of the higher-frequency signal:

    A. $f1 = 1 \div t1 = 1 \div 2 \text{ ms} = \mathbf{500\ Hz}$

    B. $f2 = 1 \div t2 = 1 \div 250\ \mu\text{s} = \mathbf{4000\ Hz}$

    C. $E1 = (E_{max} + E_{min}) \div 2 = (8 + 3) \div 2 = \mathbf{5.5\ Vpeak}$

    D. $E2 = (E_{max} - E_{min}) \div 2 = (8 - 3) \div 2 = \mathbf{2.5\ Vpeak}$

CHAPTER 10

1. B — When two signals are combined in a non-linear circuit, the output waveform will contain **both signals and intermod frequencies in addition to their harmonics**.

2. C — The frequencies created as a result of two (or more) signals combining and interacting with each other within a non-linear circuit are called **intermodulation products**.

3. C — A detailed and accurate indication of intermod signals can be obtained by using a **spectrum analyzer**.

4. Harmonics are multiples of the applied signals, while intermod products are the sum and difference of the applied signals and any harmonics. So, by deduction, you can generalize that there are a pair of intermod signals for every harmonic signal, or **fewer** harmonics.

5. D — Processing a complex waveform through a **low-pass filter** will remove all frequencies above the filter's cut-off frequency. If only the difference intermod frequency is desired, it can be best retained with a low-pass filter that removes all other frequencies from the waveform.

6. The first two intermod frequencies are the sum and difference frequencies of the applied signals.

$$f_{low} = f_2 - f_1 = 1100 - 900 = \mathbf{200\ Hz}$$

$$f_{high} = f_2 + f_1 = 1100 + 900 = \mathbf{2000\ Hz}$$

7. For this example, the intermod frequencies are:

$$f_{low} = f_2 - f_1 = 1100 - 200 = \mathbf{900\ Hz}$$

$$f_{high} = f_2 + f_1 = 1100 + 200 = \mathbf{1300\ Hz}$$

8. The third-order intermod products are the sum and difference frequencies of the second harmonics and the applied signals.

$$f_{low} = 2f_2 - f_1 = 2200 - 900 = \mathbf{1300\ Hz}$$

$$f_{high} = 2f_2 + f_1 = 2200 + 900 = \mathbf{3100\ Hz}$$

$$f_{low}' = 2f_1 - f_2 = 1800 - 1100 = \mathbf{700\ Hz}$$

$$f_{high}' = 2f_1 + f_2 = 1800 + 1100 = \mathbf{2900\ Hz}$$

9. With a passband extending from 100 to 3000 Hz, the included frequencies are:

$f_1 = 900$ Hz
$f_2 = 1100$ Hz
$2f_1 = 1800$ Hz
$2f_2 = 2200$ Hz
$3f_1 = 2700$ Hz
$f_2 - f_1 = 1100 - 900 = 200$ Hz
$f_2 + f_1 = 1100 + 900 = 2000$ Hz
$2f_1 + f_2 = 1800 + 1100 = 2900$ Hz
$2f_2 - f_1 = 2200 - 900 = 1300$ Hz
$2f_1 - f_2 = 1800 - 1100 = 700$ Hz
$3f_1 - f_2 = 2700 - 1100 = 1600$ Hz
$3f_2 - f_1 = 3300 - 900 = 2400$ Hz

There are **12 frequencies** within the filter's passband.

10. The percentage of intermodulation distortion is:

$$\% \text{ IMD} = 3 [P_{IM} \div (P_{Sig} + P_{IM})] \times 100$$

$$= 3 [0.005 \div (0.1 + 0.005)] \times 100$$

$$= \mathbf{14.3\ \%}$$

11. Often the level of specific intermod signals are compared to the level of the applied signals in the output waveform of an amplifier. The power ratio is specified in dB as:

$$dB = 10 \log (P_{IM} \div P_{SIG}) = 10 \log (2 \div 100)$$

$$= \mathbf{-17\ dB}$$

12. Referring to Figure 10-6 in the Textbook, the four third-order intermod signals are labeled 1, 6, 10, and 11. Number 11 ($2f_2 + f_1$) is the largest of these at **36 dB below** the applied output signal.

CHAPTER 11

1. A — A **linear circuit** cannot select one or more frequencies and reject and/or suppress others.

2. D — The word "carrier" refers to a signal upon which information is modulated. In this example, the information is a voice signal which is modulated upon the **carrier** frequency of a radio station.

3. The advantages of the carrier signal include its **efficiency in transmission**, and **efficiency in information capacity**.

4. D — The point of this question is to make the student aware that there are many modulation forms, only three of which are discussed in this text. The text states, however, that **at least 10** different modulation forms are used in telecommunications practice. For reference; AM, FM, PM, FSK, PSK, QAM, PAM, PDM, PPM, and PCM, are all types of modulation.

5. **Amplitude modulation** is the process by which the instantaneous voltage of the information signal controls the voltage level of the carrier signal.

6. A. USB = $f_{carrier} + f_{mod\ signal} = 60 + 8 =$ **68 kHz**

   B. LSB = $f_{carrier} - f_{mod\ signal} = 60 - 8 =$ **52 kHz**

7. A. $M_{AM} = V_{mod} \div V_c = 3 \div 5 =$ **0.6**

   B. $\%_{MAM} = (V_{mod} \div V_c)\ 100 = (3 \div 5)\ 100 =$ **60%**

8. $P_{SB} = P_c\ (M_{AM}^2 \div 2) = 2(0.5^2 \div 2) =$ **0.25 watt**

9. $P_{SB} = P_c\ (M_{AM}^2 \div 2) = 2(1^2 \div 2) =$ **1.0 watt**

   $P_T = P_c + P_{SB} = 2 + 1 =$ **3 watts**

10. With DSB-SC modulation, the carrier signal is suppressed, leaving only the **1 watt** of sideband power in the output waveform.

11. With SSB modulation, both the carrier signal and one sideband are suppressed, leaving only one sideband signal in the output waveform. Since only one of the sidebands remains, the total power in the waveform is half that of the total sideband power or **0.5 watts**.

12. The frequencies created with AM are identical to those of an unbalanced non-linear circuit.

    A. f_1 = **3 kHz**

    B. f_2 = **20 kHz**

    C. $2f_1$ = **6 kHz**

    D. $2f_2$ = **40 kHz**

    E. $3f_1$ = **9 kHz**

    F. $3f_2$ = **60 kHz**

    G. $f_2 - f_1 = 20 - 3$ = **17 kHz**

    H. $f_2 + f_1 = 20 + 3$ = **23 kHz**

    I. $2f_2 + f_1 = 40 + 3$ = **43 kHz**

    J. $2f_2 - f_1 = 40 - 3$ = **37 kHz**

    K. $3f_2 + f_1 = 60 + 3$ = **63 kHz**

    L. $3f_2 - f_1 = 60 - 3$ = **57 kHz**

13. With DSB-SC modulation, the carrier and its harmonics, along with the intermod products of the harmonics, will be removed from the modulator's output waveform.

    A. f_1 = **3 kHz**

    B. $2f_1$ = **6 kHz**

    C. $3f_1$ = **9 kHz**

    D. $f_2 - f_1 = 20 - 3$ = **17 kHz**

    E. $f_2 + f_1 = 20 + 3$ = **23 kHz**

14. In principle, the major difference between mixers and modulators is purpose. The **purpose of modulation is to use intermod frequencies to carry information; while the purpose of mixing is to change the frequency of the information being carried**.

15. A 12 kHz signal is to be up-converted to 67 kHz. This can be accomplished by mixing the 12 kHz with a LO signal so that the sum of the two signals will be 67 kHz, or by mixing the 12 kHz signal with a LO signal so that the difference of the two signals is 67 kHz.

    A. $f_{LO} = f_{out} - f_{in} = 67 - 12 =$ **55 kHz**

    B. $f_{LO} = f_{out} + f_{in} = 67 + 12 =$ **79 kHz**

16. By using a high-pass filter with a cutoff at about 65 kHz, for example, the **LO frequency (carrier) will also be removed** if it is below 65 kHz. For this example, the **55 kHz LO frequency** is the best choice; leaving only the 67 kHz in the output waveform of the mixer.

17. C — The two signals, at 58 kHz and 53 kHz respectively, are applied to a mixer. The resulting intermod products will contain the modulation of the 58 kHz signal. (Inter-modulation frequencies are created with the sideband signals as well as the carrier signal.) With a 10 kHz low-pass filter at the mixer's output, only the difference frequency, **5 kHz**, will pass through the filter. This **signal waveform will also contain the modulation that was first a part of the original 58 kHz modulated carrier signal**.

18. C — The two signals, at 105 kHz and 87 kHz, are applied to the mixer. The output of the mixer is high-pass filtered at 20 kHz, rejecting the difference-frequency intermod product. If only one frequency, the sum-frequency intermod product, appears at the filter's output, the other signals have been rejected or suppressed without filtering. The other signals include the two input signals, so we can conclude that both the input signals are suppressed in the mixer. A **double-balanced mixer** suppresses both its input signals.

Section 4

CHAPTER 12

1. A — Ribbon microphones are classified as **electromagnetic** devices.

2. C — Ceramic microphones are classified as **piezoelectric** devices.

3. D — **Carbon** microphones require an external source of current for operation.

4. C — **Condenser** microphones require an external source of voltage for operation.

5. D — **Carbon** microphones have the highest sensitivity characteristics.

6. D — **Carbon** microphones are most often used as telephone transmitters.

7. C — A reference sound pressure level of 10 microbars is equal to **84 dB SPL**.

8. C — The open-circuit output voltage for a sound pressure level of 74 dB SPL, applied to a microphone rated at –40 dB and referenced to 1 microbar, is **10 mV**.

9. B — The open-circuit output voltage for a sound pressure level of 64 dB SPL, applied to a microphone rated at –40 dB and referenced to 1 microbar, is **3.16 mV**.

10. B — The open-circuit output voltage for a sound pressure level of 64 dB SPL applied to a microphone rated at –40 dB and referenced to 10 microbars, is **1 mV**.

11. B — The sensitivity specifications for microphones and other audio transducers are referenced to a frequency of **1 kHz**.

12. A — Frequency response specifications for transducers indicate dimensions of **frequency and power**.

13. D — **Carbon** microphones produce the highest levels of electrical noise.

14. B — Higher power limits of transducers occur at signal levels that cause **3% distortion**.

15. B — For maximum power transfer from a microphone, the impedance of the circuit the microphone drives should be **equal to the microphone impedance**.

16. A — The input-to-output voltage ratios of transformers used to change impedances are equal to **the square root of their impedance ratios**.

17. A. The "mic" with the lowest impedance (**200 ohms**) has the highest power capacity.

    B. The **200-ohm** "mic" will have the higher output voltage when driving a 600-ohm circuit.

    C. The **200-ohm** "mic" will have the higher output voltage when driving a 200-ohm circuit.

18. C — Loudspeaker efficiency is a measure of the sound pressure level, in dB, that results when driving a speaker with a signal input power of **1 watt**.

19. D — Headphone efficiency is measured like loudspeaker efficiency, but a correction factor of **30 dB** is applied to compensate for the contact of the headphone with the ear.

20. The power rating of a loudspeaker is proportional to the difference between a speaker's efficiency (n) and its 3% distortion level. For this example, the **smaller** speaker has a higher power rating.

21. A microphone rated at –53 dB has an open-circuit output voltage at 53 dB below 1 volt. The calculation is:

$$V = \log_{-1} (dB/20) = \log_{-1} (-53/20) = \mathbf{2.24\ mV}$$

22. The output voltage is the same for this –53 dB specification as for the "mic" in the last problem, **2.24 mV**. Only the level of sound power required to produce this voltage is different.

23. With a sound pressure of 84 dB SPL applied, the voltage of the **condenser "mic" (of problem 22) is 2.24 mV**; while the voltage of the **dynamic "mic" (of problem 21) is 10 dB higher, at 7 mV**.

24. A. The condenser "mic" is about 1.5 dB below its –50 dB specification at 1 kHz; so its voltage is calculated based on a –51.5 dB response:

$$V = \log_{-1}(\text{dB}/20) = \log_{-1}(-51.5/20) = \mathbf{2.66\ mV}$$

B. The carbon "mic" is about equal to its –10 dB specification at 1 kHz; so its voltage is calculated based on a –10 dB response:

$$V = \log_{-1}(\text{dB}/20) = \log_{-1}(-10/20) = \mathbf{316\ mV}$$

C. The variation is from about –11 dB at 300 Hz, up to -10 dB at 1 kHz, and down to –14 dB at 3 kHz. The variation is +1 and –3 = **4 dB**.

25. The quiet speech level of 44 dB SPL is 40 dB below the "mic's" sensitivity reference of 84 dB SPL, and 30 dB below this indicates a noise level at 70 dB (30 + 40) below 1 volt:

$$V = \log_{-1}(\text{dB}/20) = \log_{-1}(-70/20) = \mathbf{0.316\ mV}$$

26. Each "mic" drives a matched impedance circuit with 3.16 mV (–50 dB) open circuit. The power capacity of each "mic" is calculated as:

A. $P_{max} = (0.5\ V_{oc})^2 \div Z = (1.58 \times 10^{-3})^2 \div 200 = \mathbf{12.5\ nW}$

$P_{dBm} = 10 \log P_{mW} = 10 \log (12.5 \times 10^{-6}) = \mathbf{-49\ dBm}$

B. $P_{max} = (0.5\ V_{oc})^2 \div Z = (1.58 \times 10^{-3})2 \div 5000 = \mathbf{0.5\ nW}$

$P_{dBm} = 10 \log P_{mW} = 10 \log (0.5 \times 10^{-6}) = \mathbf{-63\ dBm}$

27. $V_{OUT} = V_{IN}\sqrt{\left(Z_{OUT}/Z_{IN}\right)} = 2\text{ mV}\ \sqrt{(600/200)} = \mathbf{3.46\ mV}$

28. A "mic" of –54 dB sensitivity will register 2 mV on a high-impedance voltmeter, with a sound power level of 74 dB SPL. The additional 20 dB of attenuation in the meter will be added directly to the sound power level, indicating a speaker efficiency of **94 dB SPL/watt**.

29. The power required is 1 watt, times the ratio of sound power level to the speaker efficiency.

    $P = \log_{-1} [(dB\ SPL - n) \div 10]$

    $= \log_{-1} [(98 - 94) \div 10] =$ **2.5 watts**

30. A "mic" of –54 dB sensitivity will register 2 mV on a high-impedance voltmeter, with a sound power level of 74 dB SPL. The additional 13 dB of attenuation in the meter will be added directly to the sound power level; then, an additional 30 dB will be added to account for the contact between the receiver and "mic." The total will indicate a receiver (speaker) efficiency of **117 dB SPL/ watt**.

31. The sound power will be lower than the efficiency:

    $P = \log_{-1} [(dB\ SPL - n) \div 10]$

    $= \log_{-1} [(78 - 94) \div 10] =$ **25 mW**

32. The calculation is a direct power ratio:

    $P = \log_{-1} [(dB\ SPL - n) \div 10]$

    $= \log_{-1} [(105 - 82) \div 10] =$ **200 W**

CHAPTER 13

1. 1. **Power Level** (in watts, milliwatts, or dBm).

   2. **Power Gain** (in ratios or dB).

   3. **Voltage Gain** (in ratios or dB).

   4. **Frequency Response and Phase Response.**

   5. **Noise Level and Signal-to-Noise Ratios.**

   6. **Circuit Linearity and Distortion Factors.**

   7. **Signal Waveforms and Harmonics.**

   8. **Signal Bandwidth and Circuit Bandpass.**

   9. **Creation of Intermodulation Products.**

   10. **Frequency Selection and Filtering.**

2. D — The **common drain** connection has the highest input impedance and lowest output impedance.

3. A — The **NPN bipolar transistor** requires a positive voltage on two of its three terminals for operation.

4. B — The output signal is obtained from the emitter terminal when the bipolar transistor is configured as a **common collector**.

5. D — The highest power capacity, calculated as the p-p signal voltage swing to the limits of the power supply [$(P = (V_{CC})^2/(8Z_L)]$, is obtained with a **12-volt** supply and a **4-ohm** load impedance. The power is 4.5 watts.

6. C — Power capacity is increased as the square of the power supply voltage ratio.

$$| P_2/P_1 | = (V_2/V_1)^2 = (12/8)^2 = \mathbf{2.25}.$$

7. C — The highest voltage gain and power capacity, of the choices available, can be obtained by connecting a **common-emitter circuit to a common-collector circuit**.

8. D — The major trade-off advantage of the complementary-symmetry circuit over the single-transistor common-collector circuit is **power efficiency**.

9. B — **Clipping** occurs when a signal attempts to swing beyond the limits of the power supply.

10. A — **Crossover** distortion occurs when both transistors are not conducting as the signal swings across zero in a complementary-symmetry circuit.

11. C — The major advantage of the differential amplifier in amplifying low-level signals is **low noise**.

12. A — Common mode signals are **rejected** by the circuitry of the differential amplifier.

13. B — The typical gain of the differential amplifier circuit is about that of the common-emitter amplifier. Of the choices available, the correct answer is **50**.

14. B — The approximate gain of the complete amplifier circuit of Figure 4-14 is the multiple of each of the stage gains. Of the choices available, the correct answer is **2000**.

15. C — IC operational amplifiers have **differential inputs and single-ended outputs**.

16. A — The (-) input of the op-amp indicates that the signal to this terminal **is out-of-phase with the output signal**.

17. B — The voltage gain of the op-amp depends upon the ratio of the feedback resistor to the **input source resistor**.

18. C — Compared to op-amps, IC audio amplifiers have **lower gain and higher power capacity**.

19. D — If a capacitor is placed across the op-amp's feedback resistor, the **circuit gain will decrease at higher frequencies**.

20. With 8 VDC, and a collector impedance of 400 ohms, the power capacity is calculated, based on the p-p signal swing to the limits of the power supply, and the collector driving a matched load of 400 ohms.

$$P_{max} = \frac{(V_{CC})^2}{8 \times Z_{load}} = \frac{12^2}{(8 \times 400)} = \mathbf{20\ mW}$$

21. With a 12 VDC power supply, and an emitter impedance of 4 ohms to match the impedance of the speaker, the power capacity is calculated, based on the p-p signal swing to the limits of the power supply.

$$P_{max} = \frac{(V_{CC})^2}{8\ Z_{load}} = \frac{12^2}{(8 \times 4)} = \mathbf{4.5\ W}$$

22. With a "linear" amplifier, the DC power required for Problem 21 is calculated as:

$$P_{DC} = (V_{CC})^2 = \frac{12^2}{(8 \times 4)} = \mathbf{4.5\ W}$$

For "linear" operation, the DC power will be about four times the signal power.

23. With a 12 VDC power supply, the output voltage of the amplifier will swing to a 12 Vp-p value. The equivalent rms voltage is calculated as:

    Vrms = Vp-p ÷ 2.828 = 12 ÷ 2.828 = 4.24 V

    The total voltage gain is 4.24 ÷ 1 mV = **4240**.

24. With a sound power level of 90 dB SPL at 16 dB above the reference of 74 dB SPL, the microphone is delivering 1 mV at 16 dB above its reference. 1 mV is –60 dB in reference to 1 volt; and with a higher sound power level, the "mic" can be as much as 16 dB less sensitive, at **–76 dB**.

25. The voltage gain of the inverting-input op-amp is the ratio of the input to feedback resistors; so the value of the feedback resistor is the voltage gain times the input resistor value.

    $R_F = A_V R_{IN} = 800 \times 1\ k\Omega$ = **800 kΩ** .

26. The gain of the non-inverting connection of the op-amp is 1 + (R_F/R_{IN}); so with values of 15 kilohms and 1 kilohm, the voltage gain is:

$$A_V = 1 + \frac{R_F}{R_{IN}} = 1 + \frac{15\ k\Omega}{1\ k\Omega} = \mathbf{16}.$$

SECTION EXAMINATIONS AND ANSWERS

Section 1 Examination

1. What is the general word used for a component, such as a microphone or loudspeaker, that converts signals from acoustic-to-electric, or electric-to-acoustic, form?

   _____

2. Within which band of the electromagnetic spectrum are the assigned frequencies for AM broadcast radio?

   A. UHF
   B. HF
   C. MF
   D. VHF

3. The dimensions for the phase of one full wavelength are:
   _____ cycle _____deg _____ rads.

4. Signal #1 is 3 dB louder than signal #2, ther efore, signal #1 is _____ as strong as signal #2.

   A. twice
   B. three times
   C. five times
   D. ten times

5. An amplifier with 16 dB gain has a 1 mW signal applied to its input. What is the signal power at its output?

   A. 16 mW
   B. 32 mW
   C. 40 mW
   D. 48 mW

6. An attenuator reduces a 20 mW signal applied to its input to 5 mW at its output. What is its attenuation rating?

   A. 3 dB
   B. 6 dB
   C. 10 dB
   D. 15 dB

7. What is the referenced power level of a 4 mW signal?

   A. 3 dBm
   B. 4 dBm
   C. 6 dBm
   D. 8 dBm

8. What is the power level of a 10 dBm signal?

   A. 1 mW
   B. 2 mW
   C. 10 mW
   D. 20 mW

9. A transmission circuit, with RC frequency response, has a corner frequency at 3 kHz. What is the attenuation of the circuit for a signal frequency of 6 kHz?

   A. 3 dB
   B. 6 dB
   C. 9 dB
   D. 12 dB

10. What signal power is required for a 30 dB S/N, with a noise power level of 0.5 μW?

    A. 50 μW
    B. 500 μW
    C. 5 mW
    D. 50 mW

11. What is the wavelength, in meters, of a 2 kHz electrical signal? _____meters

12. What is the time period per cycle, in milliseconds, of a 2 kHz electrical signal? _____ms

13. What is the frequency of a resonant electrical network with 2 mH inductance and 2 μF capacitance? _____Hz

14. What is the signal power, in milliwatts, that exists when 6 volts rms is measured across a 600 ohm transmission circuit? _____mW

15. What is the power factor, in decimal units, of a network with 600 ohms resistance and 600 ohms reactance? _____

16. What is the true power dissipated by a resistance-capacitance network with 600 ohms resistance and 600 ohms reactance, when 6 volts rms is applied to the network?

    _____mW

17. An amplifier increases a 4 milliwatt signal at its input to 320 milliwatts at its output. What is its power gain in dB?

    _____dB

18. An amplifier has 600 ohms input and output impedances. Its input-to-output voltage gain is 200. What is its power gain in dB?
    _____dB

19. What is the power level of a 16 dBm signal? _____mW

20. What value of mismatch loss will result if a 100 ohm cable is used in a 600 ohm transmission circuit? _____dB

Section 1 Examination Answers

1. A **transducer** converts signals from acoustic-to-electric, or electric-to-acoustic, form.

2. C — The **MF** band is for AM broadcast radio.

3. One full wavelength is dimensioned as: **1** cycle, **360°**, and **2 π** rads (radians).

4. A — Signal #1 is 3 dB louder than signal #2, ther efore, signal #1 is **twice** as strong as signal #2.

5. C — An amplifier with 16 dB gain has a 1 mW signal applied to its input. The signal power at its output is **40 mW**.

6. B — An attenuator that reduces a 20 mW signal to a 5 mW signal has an attenuation rating of **6 dB**.

7. C — The referenced power level of a 4 mW signal is **6 dBm**.

8. C — The power level of a 10 dBm signal is **10 mW**.

9. C — A transmission circuit, with RC frequency response, has a corner frequency at 3 kHz. The attenuation of the circuit for a signal frequency of 6 kHz is **9 dB**.

10. B — A signal power of **500 µW** is required for a 30 dB S/N, with a noise power level of 0.5 µW.

11. The wavelength of a 2 kHz electrical signal is calculated as:

$$(300 \times 10^6 \text{ m/s}) \div 2000 \text{ Hz} = \mathbf{150{,}000 \text{ meters}}.$$

12. The time period of one cycle of a 2 kHz electrical signal is calculated as:

$$t = 1 \div 2000 \text{ Hz} = \mathbf{0.5 \text{ ms}}$$

13. The resonant frequency of a 2 mH, 2 µF network is calculated as:

$$f = 1 \div \left[2\pi\sqrt{\left(2 \times 10^{-3}\right) \times \left(2 \times 10^{-6}\right)}\right] = \mathbf{2517.7 \text{ Hz}}$$

14. The power that results from 6 Vrms applied across a 600 ohm circuit is calculated as: $P = 6^2 \div 600 =$ **60 mW**.

15. The power factor of a R-X network of 600 ohms each is calculated as:

    $\cos [\tan^{-1} (600\ \Omega \div 600\ \Omega)] =$ **0.71** (0.707)

16. The true power is calculated as:

    $P = 6^2 \div \{ 600 \cos [\tan^{-1} (600 \div 600)] \} =$ **42.4 mW**

17. The power gain of an amplifier that raises signal power from 4 mW to 320 mW is calculated as:

    $dB = 10 \log (320\ mW \div 4\ mW) =$ **19 dB**

18. The power gain of an amplifier, with equal input and output impedances, that raises signal voltage by a factor of 200 is calculated as:

    $dB = 20 \log 200 =$ **46 dB**

19. The power level of a 16 dBm signal is calculated as:

    $P_{mW} = \log_{-1} (16 \div 10) =$ **39.8 mW**

20. The mismatch loss that results from a 100 ohm cable used in a 600 ohm circuit is calculated in two steps, as:

    $Y = 600\ \Omega \div 100\ \Omega =$ **6**

    $dB = 10 \log \{1 - [(6 - 1) \div (6 + 1)]^2 \} =$ **–3.1 dB**

Section 2 Examination

1. A signal's fundamental is at 500 Hz. At what frequency is its third harmonic?

   A. 500 Hz
   B. 1000 Hz
   C. 1500 Hz
   D. 2000 Hz

2. Which waveform is **not** identified as "complex?"

   A. Sine
   B. Triangle
   C. Square
   D. Pulse

3. Which waveform results from electronic circuits that switch on and off across a charging capacitor?

   A. Sine
   B. Triangle
   C. Square
   D. Pulse

4. The significant harmonics of a 500 Hz signal extend through its seventh harmonic. What is the signal's bandwidth?

   A. 2500 Hz
   B. 3000 Hz
   C. 3500 Hz
   D. 4000 Hz

5. A complex waveform has a total power of 100 mW in the fundamental. The total power in its harmonics alone is 10 mW. What is the approximate T.H.D. of the waveform?

   A. 8%
   B. 9%
   C. 10%
   D. 11%

6. A filter, with a lower frequency limit of 500 Hz, has a bandpass of two octaves. What is its higher frequency limit?

   A. 500 Hz
   B. 1000 Hz
   C. 1500 Hz
   D. 2000 Hz

7. When using filters, the frequency range where rejection occurs is referred to as the:

   A. skirt.
   B. slope.
   C. bandpass.
   D. passband.

8. Which type of filter will pass frequencies between 3 kHz and 3.4 kHz, and reject frequencies above and below these two limits?

   A. Bandpass
   B. Bandstop
   C. Low-pass
   D. High-pass

9. A signal waveform is "distorted" in a way that increases its harmonics. What is the correct name for this kind of distortion?

   _____

10. Which harmonic numbers must be considered to determine the output filtering and input frequency limits for a ×5 frequency multiplier?

    A. Fourth and sixth
    B. Third and fourth
    C. Third and seventh
    D. Sixth and seventh

11. A 500 Hz sinewave signal (f1) with a peak voltage of 4 volts, is combined with a 1500 Hz signal (f2) with a peak voltage of 1 volt. The phase of the 1500 Hz signal crosses zero in phase with the 500 Hz signal. What are the instantaneous combined voltages at 30°, 60°, and 90° in the resulting waveform?

    _____

    _____

    _____

12. A complex waveform has a 1-watt fundamental. The levels of its four significant harmonics are 200 mW, 100 mW, 50 mW, and 20 mW respectively. Calculate the T.H.D., and record your answer as a percentage. _____%

13. What is the bandpass of a filter, in octaves, if its frequency limits are 300 Hz and 3300 Hz?

    _____octaves

14. What is the bandpass of a filter, in percentage, if its frequency limits are 1000 and 1200 Hz? _____%

15. What is the slope (selectivity) of a filter, in dB-per-octave, if its rejection is 10 dB at 800 Hz, and 20 dB at 1200 Hz? _____dB/octave

16. The components in a 1200 Hz resonant LC network have 800 ohms reactance, and 25 ohms resistance. What are the following:

    A. network Q _____

    B. network bandpass _____Hz

    C. passband frequencies (f_{hi})_____Hz to (f_{low})_____Hz

17. What is the slope (selectivity), in dB-per-octave, required for a low-pass filter to convert a square wave signal to a sine waveform with all harmonics at least 40 dB below the fundamental? _____dB/octave

18. What is the filter's passband limits for a ×5 multiplier, as required to suppress the third and seventh harmonics of input signals from 8 kHz to 10 kHz?

    _____kHz to _____kHz

19. What is the conversion efficiency of a frequency tripler that produces square waves in processing its input signals?

    Calculate your answer as a percentage and in dB.

    _____% and _____dB

20. What is the gain required of an equalizing amplifier that will bring the levels of the first three harmonics of a square waveform up to the level of the fundamental?

    _____dB

Section 2 Examination Answers

1. C — A signal's fundamental is at 500 Hz. Its third harmonic is **1500 Hz**.

2. A — A **sine** waveform is not a complex waveform.

3. B — A **triangle** waveform results from electronic circuits that switch on and off across a charging capacitor.

4. B — The significant harmonics of a 500 Hz signal extend through its seventh harmonic. The signal's bandwidth is **3000 Hz**.

5. B — The total harmonic distortion of a complex waveform with a total power of 100 mW in the fundamental, and 10 mW in its harmonics alone is calculated as:

$$\text{T.H.D.} = 10 \div (100 + 10) \times 100 = \mathbf{9\%}$$

6. D — A filter, with a lower frequency limit of 500 Hz, has a bandpass of two octaves. Its higher frequency limit is **2000 Hz**.

7. A — When using filters, the frequency range where rejection occurs is referred to as the **skirt**.

8. A — A **bandpass** filter will pass frequencies between 3 kHz and 3.4 kHz, and reject frequencies above and below these two limits.

9. **Harmonic distortion** is the name given to signal distortion that increases harmonics.

10. C — The **third and seventh** harmonics must be considered to determine the output filtering and input frequency limits for a ×5 frequency multiplier.

11. The combination of a 4-volt peak 500 Hz signal (f1), and a 1-volt peak 1500 Hz signal (f2), with in-phase zero crossing points, produces the following instantaneous values:

    $V_{if1} = 4 \sin 30° = 2V$
    $V_{if2} = 1 \sin 90° = 1V$
    $V_{itotal} = 2 + 1 =$ **3V**
    $V_{if1} = 4 \sin 60° = 3.5V$
    $V_{if2} = 1 \sin 180° = 0V$
    $V_{itotal} = 3.5 + 0 =$ **3.5V**
    $V_{if1} = 4 \sin 90° = 4V$
    $V_{if2} = 1 \sin 270° = -1V$
    $V_{itotal} = 4 + (-1) =$ **3V**

12. The T.H.D. that results from a 1-watt fundamental and a total 200 + 100 + 50 + 20 = 370 mW in harmonics is calculated as:

    T.H.D. = [0.370 ÷ (1.370)] × 100 = **27%**

13. The bandpass of a filter with frequency limits of 300 and 3300 Hz is calculated as:

    [log (3300 ÷ 300)] ÷ 0.3 = **3.47 octaves**

14. The bandpass of a filter with frequency limits of 1000 and 1200 Hz is calculated as:

    { [log (1200 ÷ 1000)] ÷ 0.3 } × 100 = **26.4%**

15. The slope of a filter in dB-per-octave is calculated by first determining the part of an octave over which data is available; then, applying ratio and proportion to the dB values as follows:

    BP = [log (1200 ÷ 800)] ÷ 0.3 = 0.59 octave
    Slope = (20 dB – 10 dB) ÷ 0.59 octave = **17 dB/octave**

16. The 1200 Hz resonant LC network, with 800 ohms reactance and 25 ohms resistance, has the following:

    A. network Q = 800 ÷ 25 = **32**
    B. network bandpass = 1200 Hz ÷ 32 = **37.5 Hz**
    C. passband frequencies:

    (f_{hi}) = 1200 Hz { 1 + [1 ÷ (2 × 32)] } = **1219 Hz**

    (f_{low}) = 1200 Hz {1 – [1 ÷ (2 × 32)] } = **1181 Hz**

17. The slope of a filter for a square wave signal, that rejects the third harmonic by 40 dB is calculated as:

    BP = (log 3) ÷ 0.3 = 1.59 octave;
    Slope = 40 dB ÷ 1.59 octave **= 25 dB/octave**

18. The passband limits of a ×5 multiplier's output filter must reject the third and seventh harmonics of the range of input frequencies. The third harmonic of the higher frequency signal must be rejected ($3 \times 10 = 30$ kHz), and the seventh harmonic of the lower frequency signal must be rejected ($7 \times 8 = 56$ kHz). Therefore, the output filter must be limited to a passband from **above 30 kHz to below 56 kHz**.

19. The third harmonic of square-wave signal distortion used in the tripler is 9 dB below (**–9 dB**) the level of the input fundamental frequency signal, as shown in Table 8-1 in the textbook. 9 dB is an 8:1 power ratio, equaling a **12.5%** conversion efficiency.

20. The first three harmonics of a square waveform are –9, –14, and –16 dB respectively, as shown in Table 8-1 in the textbook. The highest gain of an equalizing amplifier must be **16 dB** to compensate for these signal levels.

Section 3 Examination

1. What is the name of the technique used to solve electrical network problems when there are two sources of power applied to a common circuit path? _____

2. Two signals are applied across a linear T-network, with voltage and resistor values as shown in Figure 1. What is the instantaneous peak load (R3) voltage from the higher-frequency source when the lower-frequency wave is crossing zero?

   A. 0.25 Vpeak
   B. 0.5 Vpeak
   C. 1.5 Vpeak
   D. 2.0 Vpeak

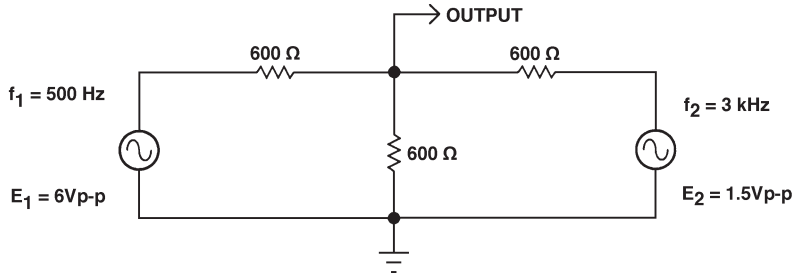

Figure 1

3. Referring to Question 2, and Figure 1, what is the instantaneous peak load (R3) voltage when both signals are at their maximum peak levels?

   A. 0.75 Vpeak
   B. 1.25 Vpeak
   C. 1.5 Vpeak
   D. 2.0 Vpeak

4. The effects of superposition, the direct adding of one signal to another, can occur only in a:

   A. balanced modulator.
   B. non-linear circuit.
   C. linear circuit.
   D. up-converter.

5. A more detailed, and accurate indication of the frequency and power of intermod signals can be obtained by using a:

   A. distortion analyzer.
   B. spectrum analyzer.
   C. oscilloscope.
   D. multimeter.

6. A circuit with an intermodulation distortion factor of less than _____ is considered *good*.

   A. 1%
   B. 5%
   C. 7%
   D. 10%

7. When the intermod sum and difference frequencies of the second harmonic of f_1 ($2f_1$) and the first order term (f_2) interact ($2f_1 \pm f_2$), the resulting pair of terms are referred to as the:

   A. second-order intermod terms.
   B. third-order harmonic terms.
   C. third-order intermod terms.
   D. fourth-order intermod terms.

8. One signal is applied to a non-linear circuit. The output will contain:

   A. the fundamental and harmonics.
   B. the fundamental only.
   C. the fundamental, harmonics, and intermods.
   D. the harmonics and intermods.

9. Two signals are applied to the input of a mixer. One signal is a modulated 67 kHz carrier, and the other is a "clean" sine wave at 53 kHz. The output signal from the mixer is low-pass filtered at 20 kHz. What is the character of the filter's output signal?

   A. An unmodulated sine wave at 14 kHz
   B. An unmodulated sine wave at 120 kHz
   C. A modulated signal at 14 kHz
   D. A modulated signal at 120 kHz

10. A local oscillator signal at 95 kHz is applied to one input terminal of a mixer. A modulated carrier signal at 107 kHz is applied to the other mixer input terminal. The output of the mixer is applied to a 50 kHz high-pass filter. The only frequency that appears at the filter's output is at 202 kHz.

    What type of mixer is used?

    A. Unbalanced
    B. Single-balanced
    C. Double-balanced
    D. Triple-balanced

11. Refer to Figure 1, and calculate the instantaneous voltage across the load (R3) when signal f1 is at its negative peak, and signal f2 is at its positive peak. _____

12. Refer to Figure 2, and calculate the values of the following. Note that f1 is the low-frequency signal, while f2 is the high-frequency signal.

    A. The frequency of the signal f1. _____

    B. The frequency of the signal f2. _____

    C. The peak voltage of the signal f1. _____

    D. The peak voltage of the signal f2. _____

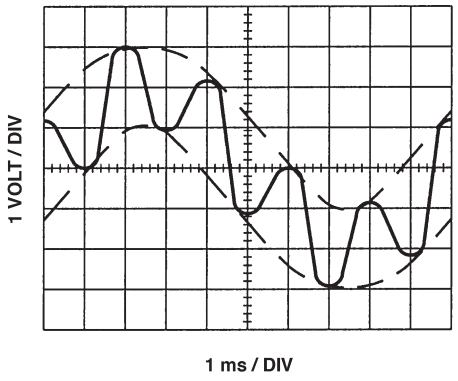

Figure 2

13. Two signals, at 1500 Hz and 2300 Hz, are processed through an unbalanced non-linear circuit. List all of the signal frequencies at the circuit's output through the third-order harmonic and intermod products. _____

_____

_____

_____

14. Two signals, at 1100 Hz and 1500 Hz, are processed through an unbalanced non-linear circuit. If the passband of the circuit is from 300 Hz through 3000 Hz, what are the harmonic and intermod frequencies that will exist within the circuit's output waveform?

_____

_____

_____

15. Two signals are applied to a non-linear circuit. The power levels of these signals are the same 1 watt at the circuit's output. The total power level of the intermod signals in the output waveform is 80 milliwatts. What is the percentage of intermodulation distortion? _____

16. Refer to Figure 3, and identify the power levels of the following if signals 3 and 4 are at 1 watt:

    A. The higher-power third-order intermod pair._____

    B. The second-order harmonic. _____

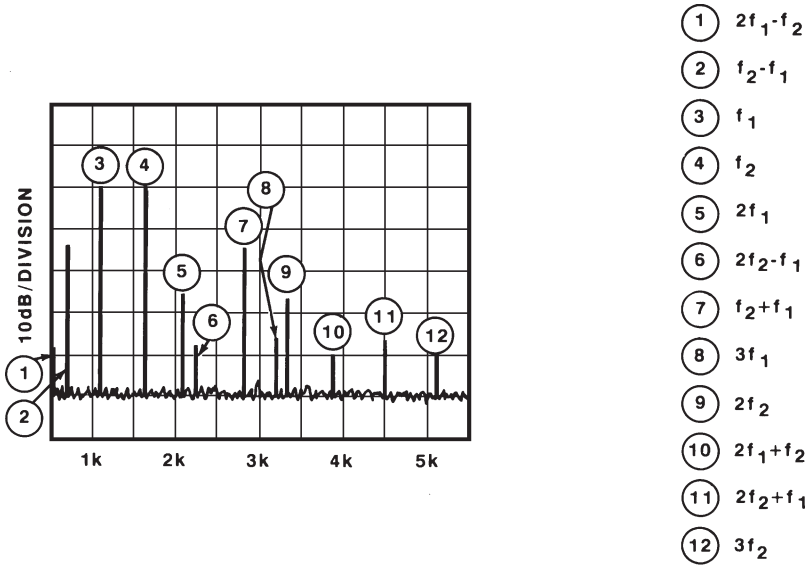

Figure 3

17. Identify which modulation form yields the following output signals:

    A. Carrier and two sidebands. _____

    B. No carrier and two sidebands. _____

    C. No carrier and one sideband. _____

18. A 4-volt voice signal amplitude-modulates a 5-volt carrier signal. The power level of the carrier signal is 1 watt.

    Calculate the following:

    A. The modulation index. _____

    B. The total sideband power. _____

19. An AM signal, with a modulation index of 100%, has a total of 3 watts of power in its carrier and sidebands. How much power is required to transmit the same information in SSB? _____

20. To up-convert a 100 kHz signal to exactly 312 kHz, which two local oscillator frequencies can be selected for the mixer input?

_____

Section 3 Examination Answers

1. **Superposition** is the name of the technique used to solve electrical network problems when there are two sources of power applied to a common circuit path.

2. A — Calculate as follows:

$$E_{R3}' = E_{G2} [(R1 \,||\, R3) \div (R2 + R1 \,||\, R3)]$$

$$= 1.5 [300 \div 600 + 300] = 0.5 \text{ Vp-p}$$

$$= \mathbf{0.25 \text{ Vpeak}}$$

3. B — Calculate as follows:

$$E_{R3} = E_{G1} [(R2 \,||\, R3) \div (R1 + R2 \,||\, R3)]$$

$$= 6 [300 \div 600 + 300] = 2.0 \text{ Vp-p}$$

$$= 1.0 \text{ Vpeak}$$

E_{R3}' is 0.25 V from answer to Question 2;

$E_{R3} + E_{R3}' = 1.0 + 0.25$ **= 1.25 Vpeak**

4. C — The effects of superposition—the direct adding of one signal to another—can occur only in a **linear circuit**.

5. B — A more detailed, and accurate indication of the frequency and power of intermod signals can be obtained by using a **spectrum analyzer**.

6. A — A circuit with an intermodulation distortion factor of less than **1%** is considered good.

7. C — When the intermod sum and difference frequencies of the second harmonic of f_1 ($2f_1$) and the first order term (f_2) interact ($2f_1 \pm f_2$), the resulting pair of terms are referred to as the **third-order intermod terms**.

8. A — One signal is applied to a non-linear circuit. The output will contain **the fundamental and harmonics**.

9. C — Two signals are applied to the input of a mixer. One signal is a modulated 67 kHz carrier, and the other is a "clean" sine wave at 53 kHz. The output signal from the mixer is low-pass filtered at 20 kHz. The filter's output signal is a **modulated signal at 14 kHz**.

10. C — A local oscillator signal at 95 kHz is applied to one input terminal of a mixer. A modulated carrier signal at 107 kHz is applied to the other mixer input terminal. The output of the mixer is applied to a 50 kHz high-pass filter. The only frequency that appears at the filter's output is at 202 kHz. A **double-balanced** mixer is used.

11. Calculate as follows:

$$E_{R3}' = E_{G2} [(R1 \,||\, R3) \div (R2 + R1 \,||\, R3)]$$

$$= 1.5 [300 \div (600 + 300)] = 0.5 \text{ Vp-p}$$

$$= 0.25 \text{ Vpeak}$$

$$E_{R3} = E_{G1} [(R2 \,||\, R3) \div (R1 + R2 \,||\, R3)]$$

$$= 6 [300 \div (600 + 300)] = 2.0 \text{ Vp-p}$$

$$= 1.0 \text{ Vpeak}$$

$$-E_{R3} + E_{R3}' = -1.0 + 0.25 = \mathbf{-0.75 \text{ Vpeak}}$$

12. A. The frequency of f1 is **100 Hz**, based on one cycle of the signal occupying 10 milliseconds. $f = 1/t$

    B. The frequency of f2 is **500 Hz**, based on one cycle of the signal occupying 2 milliseconds. $f = 1/t$

    C. $E_{f1} = (E_{max} + E_{min}) \div 2 = (3 + 1) \div 2 = \mathbf{2 \text{ Vpeak}}$

    D. $E_{f2} = (E_{max} - E_{min}) \div 2 = (3 - 1) \div 2 = \mathbf{1 \text{ Vpeak}}$

13. There are twelve frequencies, calculate as follows:

f_1 = **1500 Hz**
f_2 = **2300 Hz**
$2f_1$ = **3000 Hz**
$2f_2$ = **4600 Hz**
$3f_1$ = **4500 Hz**
$3f_2$ = **6900 Hz**
$f_2 - f_1 = 2300 - 1500 =$ **800 Hz**
$f_2 + f_1 = 2300 + 1500 =$ **3800 Hz**
$2f_1 + f_2 = 3000 + 2300 =$ **5300 Hz**
$2f_1 - f_2 = 3000 - 2300 =$ **700 Hz**
$2f_2 + f_1 = 4600 + 1500 =$ **5100 Hz**
$2f_2 - f_1 = 4600 - 1500 =$ **3100 Hz**

14. There are eight frequencies, calculate as follows:

f_1 = **1100 Hz**
f_2 = **1500 Hz**
$2f_1$ = **2200 Hz**
$2f_2$ = **3000 Hz**
$f_2 - f_1 = 1500 - 1100 =$ **400 Hz**
$f_2 + f_1 = 1500 + 1100 =$ **2600 Hz**
$2f_1 - f_2 = 2200 - 1500 =$ **700 Hz**
$2f_2 - f_1 = 3000 - 1500 =$ **1500 Hz**

15. The percentage of IMD is calculated as follows:

$$\%IMD = 3 [P_{IM} \div (P_{SIG} + P_{IM})] \times 100$$

$$= 3 [0.08 \div (1 + 0.08)] \times 100$$

$$=$$ **22.2%**

16. A. The higher-powered third-order intermod pair are numbered 6 and 11. These signals are at **0.2 mW**, 37 dB below the input signals 3 and 4.

    B. The second-order harmonic signals are numbered 5 and 9. These signals are at **2.5 mW**, 26 dB below the input signals. The power levels are calculated using the inverse log formula from Section 1.

17. A. Carrier and two sidebands. **AM (DSB-AM)**

    B. No carrier and two sidebands. **Double-sideband suppressed-carrier (DSB-SC)**

    C. No carrier and one sideband. **Single sideband (SSB)**

18. A 4-volt voice signal amplitude-modulates a 5-volt carrier signal. The power level of the carrier signal is 1 watt.

    A. $M_{AM} = V_{mod} \div V_c = 4 \div 5 =$ **0.8 or 80%**

    B. $P_{SB} = P_c (M_{AM}^2 \div 2) = 1 (0.8^2 \div 2) =$ **0.32 watt**

19. With a modulation index of 1, the power in the sidebands of an AM signal is half the carrier power—the total power is 1.5 times the carrier power. With a total power of 3 watts, there will be 2 watts in the carrier, and 1 watt in both sidebands. With SSB modulation the power in one sideband is half that of both sidebands, therefore, the same amount of information can be transmitted with **0.5 watt** in SSB, compared to 3 watts in AM!

20. A 100 kHz signal can be up-converted to exactly 312 kHz by using a **212 kHz or 412 kHz** local oscillator frequency.

Section 4 Examination

1. Which of the following microphones are most often used as telephone transmitters?

   A. Dynamic
   B. Crystal
   C. Carbon
   D. Condenser

2. A microphone's output is listed as –40 dB, referenced to 0 dB = 1V/microbar. What is its open-circuit output voltage when a sound pressure level of 64 dB SPL is applied to its input?

   A. 1 mV
   B. 3.16 mV
   C. 10 mV
   D. 31.6 mV

3. Which of the following microphone types produces the highest levels of electrical noise?

   A. Dynamic
   B. Crystal
   C. Carbon
   D. Condenser

4. The higher power limits of transducers occur at signal levels that cause:

   A. less than 1% distortion.
   B. 1% distortion.
   C. 3% distortion.
   D. burn-out.

5. A large loudspeaker is 14 dB more efficient than a small loudspeaker. The small speaker reaches its 3% distortion level at 25 dB above its efficiency value (n); while the large speaker reaches its 3% distortion level at 22 dB above its efficiency value. Which speaker will have the higher power rating, large or small? _____

6. A dynamic microphone is specified for sensitivity at –52 dB, with 0 dB = 1 volt/microbar. What is its open-circuit output voltage level? _____

7. A test circuit is connected to measure the efficiency of a telephone receiver. The output from the standard microphone in contact with the receiver reads 2 mV on a high impedanc voltmeter with 16 dB set into the meter's attenuator. What is the receiver's efficiency, in dB SPL/watt? _____

8. A loudspeaker with an efficiency of 85 dB SPL reaches 3% distortion when it is driven to 105 dB SPL. What is its maximum power rating? _____

9. In which transistor circuit connection is the output signal obtained from the emitter terminal?

   A. Common source
   B. Common drain
   C. Common emitter
   D. Common collector

10. What is the major trade-off advantage of the two-transistor complementary-symmetry circuit over the single-transistor common-collector circuit?

    A. Power capacity
    B. Power efficiency
    C. Voltage gain
    D. Current gain

11. The typical voltage gain of the differential amplifier circuit is about:

    A. 6.
    B. 60.
    C. 600.
    D. 6000.

12. With a 12 VDC power supply, and a collector impedance of 600 ohms, what is the power capacity of a transistor audio amplifier stage? _____mW

13. With a 16-VDC power supply, what is the maximum power available to an 8-ohm loudspeaker, driven by a matched-impedance common-collector transistor audio amplifier stage? _____watts

14. A microphone has a maximum rms output voltage of 4 millivolts. It is desired to raise that signal to drive a 4-ohm loudspeaker to maximum power with a 12-VDC power upply. What is the total voltage gain required of the amplifier circuit? _____

15. An inverting-input op-amp circuit has a voltage gain of 500. Its input resistor is 2 kilohms. What is the value of its feedback resistor? _____

16. A non-inverting op-amp circuit has an input resistor of 2 kilohms, and a feedback resistor of 20 kilohms. What is the circuit's voltage gain? _____

17. A microphone intended for use in an FM radio broadcast studio has a _____ range of frequency response than a microphone intended for use in a telephone transmitter.

    A. wider
    B. narrower

18. What components are often used to match microphone impedances to cable impedances to reduce the total system losses?

    A. Transducers
    B. Transformers
    C. Amplifiers
    D. Loudspeakers

19. Which piece of test equipment is used to measure speaker efficiency most practically?

    A. RMS voltmeter
    B. Oscilloscope
    C. Spectrum analyzer
    D. Distortion analyzer

20. A reference sound pressure level equal to 1 microbar is:

    A. 94 dB SPL.
    B. 84 dB SPL.
    C. 74 dB SPL.
    D. 64 dB SPL.

Section 4 Examination Answers

1. C — A **carbon** microphone is most often used as a telephone transmitter.

2. B — A microphone's output is listed as –40 dB, referenced to 0 dB = 1V/microbar. Its open-circuit output voltage, when a sound pressure level of 64 dB SPL is applied to its input, is **3.16 mV**.

3. C — A **carbon** microphone produces the highest levels of electrical noise.

4. C — The higher power limits of transducers occur at signal levels that cause **3% distortion**.

5. The **smaller** speaker has the higher power rating, as the power is based on the ratio of its efficiency to its 3% distortion level.

6. 52 dB below 1 volt is: $\log_{-1}$ (–52/20) = **2.5 mV.**

7. 2 mV equals the 74 dB SPL reference with a standard microphone; so 16 dB above that reference, plus 30 dB for the "contact compensation," is an efficiency of: 74 + 16 + 30 = **120 dB SPL/watt.**

8. The power rating is 20 dB (105 dB SPL – 85 dB SPL) above the 1-watt reference. P = $\log_{-1}$ (dB SPL – n) ÷ 10 = $\log_{-1}$ (105 – 85) ÷ 10 = **100 watts.**

9. D — In a **common collector** transistor circuit connection, the output signal is obtained from the emitter terminal.

10. B — **Power efficiency** is the major trade-off advantage of the two-transistor complementary-symmetry circuit over the single-transistor common-collector circuit.

11. B — The typical voltage gain of the differential amplifier circuit is about **60**.

12. $P_{MAX} = 12^2 \div (8 \times 600)$ = **30 mW.**

13. $P_{MAX} = 16^2 \div (8 \times 8)$ = **4 W**.

14. Vrms = 12 Vp-p ÷ 2.828 = 4.24 Vrms

    A_V = 4.24 V ÷ 4 mV = **1060**

15. $R_F = A_V R_{IN} = 500 \times 2\ k\Omega =$ **1 M Ω**

16. $A_V = 1 + (R_F \div R_{IN}) = 1 + (20\ k\Omega \div 2\ k\Omega) =$ **11**

17. A — A microphone intended for use in an FM radio broadcast studio has a **wider** range of frequency response than a microphone intended for use in a telephone transmitter.

18. B — **Transformers** are often used to match microphone impedances to cable impedances to reduce the total system losses.

19. A — An **rms voltmeter** is used to measure speaker efficiency most practically.

20. C — A reference sound pressure level equal to 1 microbar is **74 dB SPL**.

FINAL EXAMINATION

1. Which band of the electromagnetic spectrum has signals with the longest wavelengths?

   A. UHF
   B. HF
   C. MF
   D. VHF

2. What is the wavelength of a 220 Hz acoustic sound in feet?

   A. 1.36×10^6
   B. 1.5
   C. 4.47×10^6
   D. 5

3. Electrical noise will interfere most with signals when the noise is:

   A. distributed over a range of frequencies.
   B. near the signal's frequency.
   C. near the signal's power level.
   D. at varying power levels.

4. Two conditions are required for accurate power ratings calculated from sine wave signals. The signal voltage and current must be:

   A. in rms units, and 90° phase.
   B. in p-p units, and 90° phase.
   C. in rms units, and 0° phase.
   D. in p-p units, and 0° phase.

5. Which power rating is most often applied to signals rather than components?

   A. Power level
   B. Power gain
   C. Power loss
   D. Power consumption

6. Which impedance level is most often used for voice signal transmission circuits?

   A. 600 ohms
   B. 300 ohms
   C. 75 ohms
   D. 50 ohms

7. Which type of network will consume the maximum true power applied to it?

   A. Reactive (inductive or capacitive)
   B. Resistance-capacitance (RC)
   C. Resistance-inductance (RL)
   D. Completely resistive

8. In what units are power gains and losses measured?

   A. Factors
   B. Watts
   C. Volts
   D. Ohms

9. What is the equivalent gain for a power ratio of 20:1?

   A. 1.3 dB
   B. 4 dB
   C. 13 dB
   D. 20 dB

10. An amplifier with 13 dB gain has a 1 mW signal applied to its input. What is the signal power at its output?

    A. 13 mW
    B. 16 mW
    C. 20 mW
    D. 26 mW

11. An attenuator reduces a 40 mW signal applied to its input, to 20 mW at its output. What is its attenuation rating?

    A. 3 dB
    B. 6 dB
    C. 10 dB
    D. 15 dB

12. What is the referenced power level of an 8 mW signal?

    A. 3 dBm
    B. 4 dBm
    C. 6 dBm
    D. 9 dBm

13. What is the power level of a 13 dBm signal?

    A. 1 mW
    B. 2 mW
    C. 10 mW
    D. 20 mW

14. Which quantity is not measured in dB?

    A. Signal power level
    B. Sound power level
    C. Amplifier gain
    D. Path loss

15. At what approximate frequency is the human ear least sensitive to soft (quiet) sound levels?

    A. 100 Hz
    B. 400 Hz
    C. 1000 Hz
    D. 4000 Hz

16. What information about a signal is available from measurements using the oscilloscope, but not from the spectrum analyzer?

    A. Voltage
    B. Power
    C. Frequency
    D. Time period

17. Two sine waves are combined, with one at a frequency three times higher than the other. At what angle of rotation is the resulting waveform at maximum voltage?

    A. 0°
    B. 30°
    C. 60°
    D. 90°

18. A signal's fundamental is at 600 Hz. At what frequency is its third harmonic?

    A. 600 Hz
    B. 1200 Hz
    C. 1800 Hz
    D. 2400 Hz

19. Which harmonic number has the highest power level in a square waveform?

    A. Second
    B. Third
    C. Fourth
    D. Fifth

20. A rectified sine waveform includes its first two higher-power harmonics within its waveform. What is the highest harmonic number in the waveform?

    A. Second
    B. Third
    C. Fourth
    D. Fifth

21. The significant harmonics of a 500 Hz signal extend through its sixth harmonic. What is the signal's bandwidth?

    A. 2500 Hz
    B. 3000 Hz
    C. 3500 Hz
    D. 4000 Hz

22. A 500 Hz pulse waveform is applied to a circuit with a passband beginning at 450 Hz, and extending through 2550 Hz. What is the highest harmonic number of this waveform that will pass through the circuit?

    A. Second
    B. Third
    C. Fourth
    D. Fifth

23. The total power of a complex waveform is 400 mW. The total power in its harmonics alone is 40 mW. What is the approximate T.H.D. of the waveform?

    A. 8%
    B. 9%
    C. 10%
    D. 11%

24. A signal waveform is "distorted" in a way that reduces its harmonics. What is the correct name for this kind of distortion?

    A. Waveform distortion
    B. Intermodulation
    C. Harmonic distortion
    D. Filtering

25. A filter has a passband from 500 Hz to 1500 Hz. What is its bandpass?

    A. 500 Hz
    B. 1000 Hz
    C. 1500 Hz
    D. 2000 Hz

26. Which type of filter will pass frequencies above 5 kHz, but reject frequencies below 5 kHz?

    A. Bandpass
    B. Bandstop
    C. Low-pass
    D. High-pass

27. Which type of filter will pass frequencies between 2 kHz and 2.4 kHz, and reject frequencies above and below these two limits?

    A. Bandpass
    B. Bandstop
    C. Low-pass
    D. High-pass

28. Which type of filtering will result from a series LC resonant network placed across (in parallel, or shunt to ground) a circuit?

    A. Bandpass
    B. Bandstop
    C. Low-pass
    D. High-pass

29. A larger bandwidth (bandpass) in an LC resonant network will result from:

    A. reduced Q and higher selectivity.
    B. reduced Q and lower selectivity.
    C. increased Q and higher selectivity.
    D. increased Q and lower selectivity.

30. Which signal waveform will provide a higher-power third harmonic, and lower-power second and fourth harmonics for use within the circuitry of a frequency tripler?

    A. Pulse
    B. Square
    C. Triangle
    D. Rectified sine

31. What is the ideal waveform for a frequency doubler?

    A. Pulse
    B. Square
    C. Triangle
    D. Rectified sine

32. When two signals are applied to a linear circuit, the output waveform will contain:

    A. the two signals and their harmonics.
    B. the two signals, their harmonics, and intermod frequencies.
    C. the two signals only.
    D. harmonics and intermod frequencies only.

33. When two signals are applied to an unbalanced non-linear circuit, the output waveform will contain:

    A. the two signals and their harmonics.
    B. the two signals, their harmonics, and intermod frequencies
    C. the two signals only.
    D. harmonics and intermod frequencies only.

34. When two signals are applied to a balanced non-linear circuit, which signals will be suppressed in the output waveform?

    A. The two signals and their harmonics.
    B. The two signals, their harmonics, and intermod frequencies.
    C. The two signals only.
    D. Harmonics and intermod frequencies only.

35. When two signals are applied to an unbalanced non-linear circuit, and harmonics and intermod products extend through their third-order values, how many intermod frequencies will exist?

    A. 3
    B. 6
    C. 9
    D. 12

36. Which circuit cannot be used to select and/or suppress specific harmonic or intermod frequencies in the output waveform of a complex signal?

    A. Linear circuit
    B. Filter circuit
    C. Phase-cancelling circuit
    D. Balanced modulator circuit

37. Two signals are applied to the input of a mixer. One signal is a modulated 57 kHz carrier, and the other is a "clean" sine wave at 43 kHz. The output signal from the mixer is low-pass-filtered at 20 kHz. What is the character of the filter's output signal?

    A. An unmodulated sine wave at 14 kHz.
    B. An unmodulated sine wave at 100 kHz.
    C. A modulated signal at 14 kHz.
    D. A modulated signal at 100 kHz.

38. Two signals, at 1500 Hz and 2300 Hz, are processed through an unbalanced non-linear circuit. How many signal frequencies are at the circuit's output, through the third-order harmonic and intermod products?

    A. 4
    B. 8
    C. 12
    D. 18

39. Two signals, at 1100 Hz and 1500 Hz, are processed through an unbalanced non-linear circuit. If the passband of the circuit is from 300 Hz through 3000 Hz, how many harmonic and intermod frequencies will exist within the circuit's output waveform?

    A. 4
    B. 8
    C. 12
    D. 18

40. Two signals are applied to a non-linear circuit. The power levels of these signals are the same 1 watt at the circuit's output. The total power level of the intermod signals in the output waveform is 80 milliwatts. What is the percentage of intermodulation distortion?

    A. 11%
    B. 22%
    C. 33%
    D. 44%

41. Which modulation form yields no carrier, but two sidebands?

    A. DSB-AM
    B. VSB
    C. SSB
    D. DSB-SC

42. Which modulation form yields no carrier and one sideband?

    A. DSB-AM
    B. VSB
    C. SSB
    D. DSB-SC

43. A 4-volt voice signal amplitude-modulates a 5-volt carrier signal. What is the modulation index?

    A. 20%
    B. 40%
    C. 60%
    D. 80%

44. A 4-volt voice signal amplitude-modulates a 5-volt carrier signal. The power of the carrier signal is 1 watt. What is the total sideband power?

    A. 0.16 W
    B. 0.32 W
    C. 0.48 W
    D. 0.64 W

45. A multiplexed voice signal is modulated on a 63 kHz carrier. That signal is applied to a mixer, to be up-converted to 88 kHz. Which LO frequency can be selected for the mixer input?

    A. 25 kHz
    B. 63 kHz
    C. 88 kHz
    D. 100 kHz

46. Which of the following microphones require an external source of current for operation?

    A. Dynamic
    B. Crystal
    C. Carbon
    D. Condenser

47. A reference sound pressure level equal to 1 microbar is:

    A. 64 dB SPL.
    B. 74 dB SPL.
    C. 84 dB SPL.
    D. 94 dB SPL.

48. Sensitivity specifications for transducers are referenced to a frequency of:

    A. 300 Hz.
    B. 1 kHz.
    C. 3 kHz.
    D. 10 kHz.

49. When transformers are used to change transducer impedances, their input-to-output voltage ratios are equal to:

    A. one-half their impedance ratios.
    B. the square root of their impedance ratios.
    C. their impedance ratios.
    D. the square of their impedance ratios.

50. A high-output carbon microphone has a measured sensitivity of –20 dB, with a reference of 1 volt/10 microbars (84 dB SPL). What is its open-circuit output voltage level?

    A. 0.001 volt
    B. 0.01 volt
    C. 0.1 volt
    D. 1.0 volt

51. Two microphones are rated at the same sensitivity, of –46 dB. The impedance of one "mic" is 200 ohms, while the impedance of the other is 5 kilohms. Which of the "mics" have the higher power capacity?

    A. 200 ohm
    B. 5 kilohm

52. A test circuit is connected to measure the efficiency of a loudspeaker. The output from the standard microphone reads 2 mV on a high impedance voltmeter with 26 dB set into the meter's attenuator. What is the efficiency of the speaker, in dB SPL/watt?

    A. 70 dB SPL/watt
    B. 80 dB SPL/watt
    C. 90 dB SPL/watt
    D. 100 dB SPL/watt

53. Which transistor circuit connection has the highest input and lowest output impedances?

    A. Common source
    B. Common drain
    C. Common emitter
    D. Common collector

54. If the power supply voltage to a transistor is increased from 10 VDC to 14 VDC, by what factor will the transistor's power capacity increase?

    A. 2.0
    B. 2.4
    C. 3.6
    D. 4.0

55. The highest voltage gain and power capacity can be obtained by connecting which two stages together?

    A. A common-emitter driving a common-collector.
    B. A common-collector driving a common-emitter.
    C. A common-emitter driving a common-emitter.
    D. A common-collector driving a common-collector.

56. What is the name given to the waveform distortion that occurs when the signal swings beyond the limits of a device's power supply?

    A. Harmonic
    B. Crossover
    C. Clipping
    D. Intermodulation

57. What is the major advantage of the differential amplifier in amplifying low-level signals?

    A. High gain
    B. Low noise
    C. Low distortion
    D. High power capacity

58. IC operational amplifiers have:

    A. differential inputs and differential outputs.
    B. single-ended inputs and differential outputs.
    C. differential inputs and single-ended outputs.
    D. single-ended inputs and single-ended outputs.

59. The voltage gain of an operational amplifier circuit depends upon the ratio of the feedback resistor to the:

    A. input source resistor.
    B. output load resistor.
    C. internal source resistance of the IC.
    D. internal load resistance of the IC.

60. Placing a capacitor across the feedback resistor in an op-amp circuit will:

    A. increase the circuit gain at high frequencies.
    B. increase the circuit gain at low frequencies.
    C. decrease the circuit gain at low frequencies.
    D. decrease the circuit gain at high frequencies.

FINAL EXAMINATION ANSWERS

1. C — The **MF** band of the electromagnetic spectrum has signals with the longest wavelengths.

2. D — The wavelength of a 220 Hz acoustic sound is **5 feet**.

3. C — Electrical noise will interfere most with signals when the noise is **near the signal's power level**.

4. C — Two conditions are required for accurate power ratings calculated from sine wave signals. The signal voltage and current must be **in rms units, and 0° phase**.

5. A — The power rating most often applied to signals rather than components is **power leve**l.

6. A — The impedance level most often used for voice signal transmission circuits is **600 ohms**.

7. D — A **completely resistive** network will consume the maximum true power applied to it.

8. A — Power gains and losses are measured in **factors**.

9. C — A power ratio of 20:1 is equivalent to a gain of **13 dB**.

10. C — An amplifier with 13 dB gain and a 1 mW signal applied to its input has a signal power of **20 mW** at its output.

11. A — An attenuator reduces a 40 mW signal applied to its input, to 20 mW at its output. Its attenuation rating is **3 dB**.

12. D — **9 dBm** is the referenced power level of an 8 mW signal.

13. D — **20 mW** is the power level of a 13 dBm signal.

14. A — **Signal power level** is not measured in dB.

15. A — At approximately **100 Hz**, the human ear is least sensitive to soft (quiet) sound levels.

16. D — **Time period** of a signal is shown in measurements using the oscilloscope, but not using the spectrum analyzer.

17. B — Two sine waves are combined, with one at a frequency three times higher than the other. The waveform is at maximum voltage at a **30°** angle of rotation.

18. C — If a signal's fundamental is at 600 Hz, its third harmonic is at **1800 Hz**.

19. B — The **third** harmonic number has the highest power level in a square waveform.

20. C — A rectified sine waveform includes its first two higher-power harmonics within its waveform. The highest harmonic number in the waveform is the **fourth**.

21. A — The bandwidth of a 500 Hz signal, whose significant harmonics extend through its sixth harmonic, is **2500 Hz**.

22. D — A 500 Hz pulse waveform is applied to a circuit with a passband beginning at 450 Hz, and extending through 2550 Hz. The **fifth** harmonic of this waveform is the highest that will pass through this circuit.

23. B — The total power of a complex waveform is 400 mW. The total power in its harmonics alone is 40 mW. The approximate T.H.D. of the waveform is **9%**.

24. D — **Filtering** distorts a waveform in a way that reduces its harmonics.

25. B — A filter with a passband from 500 Hz to 1500 Hz has a bandpass of **1000 Hz**.

26. D — A **high-pass** filter will pass frequencies above 5 kHz, but reject frequencies below 5 kHz.

27. A — A **bandpass** filter will pass frequencies between 2 kHz and 2.4 kHz, and reject frequencies above and below these two limits.

28. B — **Bandstop** filtering will result from a series LC resonant network placed across a circuit.

29. B — A larger bandwidth (bandpass) in an LC resonant network will result from **reduced Q and lower selectivity**.

30. B — The **square** waveform will provide a higher-power third harmonic, and lower-power second and fourth harmonics for use within the circuitry of a frequency tripler.

31. D — The ideal waveform for a frequency doubler is a **rectified sine**.

32. C — When two signals are applied to a linear circuit, the output waveform will contain **the two signals only**.

33. B — When two signals are applied to an unbalanced non-linear circuit, the output waveform will contain **the two signals, their harmonics, and intermod frequencies**.

34. A — When two signals are applied to a balanced non-linear circuit, **the two signals and their harmonics** will be suppressed in the output waveform.

35. D — When two signals are applied to an unbalanced non-linear circuit, and harmonics and intermod products extend through their third-order values, **12** intermod frequencies will exist.

36. A — **A linear circuit** cannot be used to select and/or suppress specific harmonic or intermod frequencies in the output waveform of a complex signal.

37. C — Two signals are applied to the input of a mixer. One signal is a modulated 57 kHz carrier, and the other is a "clean" sine wave at 43 kHz. The output signal from the mixer is low-pass-filtered at 20 kHz. The filter's output is a **modulated signal at 14 kHz**.

38. C — Two signals, at 1500 Hz and 2300 Hz, are processed through an unbalanced non-linear circuit. **Twelve** signal frequencies, through the third-order harmonic and intermod products, are at the circuit's output.

39. B — Two signals, at 1100 Hz and 1500 Hz, are processed through an unbalanced non-linear circuit. If the passband of the circuit is from 300 Hz through 3000 Hz, **eight** harmonic and intermod frequencies will exist within the circuit's output waveform.

40. B — Two signals are applied to a non-linear circuit. The power levels of these signals are the same 1 watt at the circuit's output. The total power level of the intermod signals in the output waveform is 80 milliwatts. The percentage of intermodulation distortion is **22%**.

41. D — **DSB-SC** yields no carrier, but two sidebands.

42. C — **SSB** yields no carrier and one sideband.

43. D — A 4-volt voice signal amplitude-modulates a 5-volt carrier signal. The modulation index is **80%**.

44. B — A 4-volt voice signal amplitude-modulates a 5-volt carrier signal. The power of the carrier signal is 1 watt. The total sideband power is **0.32 W**.

45. A — A multiplexed voice signal is modulated on a 63 kHz carrier. That signal is applied to a mixer, to be up-converted to 88 kHz. A **25 kHz** LO frequency can be used for the mixer input.

46. C — A **carbon** microphone requires an external source of current for operation.

47. B — A reference sound pressure level equal to 1 microbar is **74 dB SPL**.

48. B — Sensitivity specifications for transducers are referenced to a frequency of **1 kHz**.

49. B — When transformers are used to change transducer impedances, their input-to-output voltage ratios are equal to **the square root of their impedance ratios**.

50. C — A high-output carbon microphone has a measured sensitivity of –20 dB, with a reference of 1 volt/10 microbars (84 dB SPL). Its open-circuit output voltage level is **0.1 volt**.

51. A — Two microphones are rated at the same sensitivity, of –46 dB. The impedance of one "mic" is 200 ohms, while the impedance of the other is 5 kilohms. The **200 ohm** "mic" has the highest power capacity.

52. D — A test circuit is connected to measure the efficiency of a loudspeaker. The output from the standard microphone reads 2 mV on a high impedance voltmeter with 26 dB set into the meter's attenuator. The efficiency of the speaker is **100 dB SPL/watt**.

53. B — The **common drain** transistor circuit connection has the highest input and lowest output impedances.

54. A — If the power supply voltage to a transistor is increased from 10 VDC to 14 VDC, the transistor's power capacity will increase by a factor of **2.0**.

55. A — The highest voltage gain and power capacity can be obtained by connecting **a common-emitter driving a common-collector**.

56. C — When the signal swings beyond the limits of a device's power supply, the waveform distortion that occurs is called **clipping**.

57. B — The major advantage of the differential amplifier in amplifying low-level signals is **low noise**.

58. C — IC operational amplifiers have **differential inputs and single-ended outputs**.

59. A — The voltage gain of an operational amplifier circuit depends upon the ratio of the feedback resistor to the **input source resistor**.

60. D — Placing a capacitor across the feedback resistor in an op-amp circuit will **decrease the circuit gain at high frequencies**.